





XML / JSON API Reference

MAINTENANCE RELEASE 80



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Table of Contents

	Security	
	•	
^	ML API	
	Error Handling	
1	SON API	
J	Access to JSON API	
	Error handling	
۷	VSDL	
ı	Reference	11
١	lotation conventions	12
E	stablishing an authenticated session	12
	Methods	12
	Type reference	13
C	Global methods and types	
	Type reference	13
٧	oicemail Settings	14
	Methods	14
	Type reference	
F	folder preferences and Mailbox and message display options	
	Methods	
_	Type reference	
Α	uto attendant configuration	
	Methods	
_	Type reference	
C	Conference configuration	
	Methods	
	Type reference	
	Call control API	
C	Overview	
	Access to JSON-RPC API	
_	Error handling	
C	Call state notification management	
	Methods	
	Type reference	
٧	oice API	
	Methods	
_	Type Reference	
C	Call control API for IVR	
	Methods	
_	Type Reference	
C	Conferencing API	
	Methods Type reference	
	Appendices	95



Appendix A. Sample script for PortaSIP® media server SOAP	
communication	96
Appendix B. Call control API usage examples	99
JavaScript	99
Python	
Appendix C. How to define a time period	



Preface

This document provides information for developers who want to interface their applications with PortaSIP® media applications via XML and JSON API. The PortaBilling® XML and JSON API is described in the **PortaBilling XML / JSON API Reference**.

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 that occur inbetween minor 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:



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.

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1 - XML / JSON API Overview



Security

Connection to the XML / JSON API interface is provided via HTTPS. Authentication is done using authentication pairs (login-password or login-API token). Each subsequent request to the API should contain the **auth info** structure.



Note that we strongly recommend using the <code>session_id</code> property (which is received during the authorization via the <code>LoginRequest</code>) in the <code>auth_info</code> structure for all session requests. Otherwise, if you use login-password authentication pairs for every request, new sessions will be created and will cause additional load to the database.

XML API

XML (SOAP) API has its own advantages and drawbacks as compared with JSON API. Among the benefits are the following:

- There is a wide range of reusable software available to programmers to handle XML so they do not have to re-invent code.
- XML (SOAP) is more verbose compared with JSON, but because
 of this, the data encoding result is typically larger than the
 equivalent encoding in JSON API.

Access to XML API

Proxy (URL): https://web-server.yourdomain.com:8443/soap/soap.fcgi SOAP URI (namespace): https://web-server.yourdomain.com/UM/SOAP/



Please replace the **web-server.yourdomain.com** with the actual hostname of your PortaSwitch® web server.

Error Handling

SOAP faults are used to carry error information within a SOAP message. If the actual response has a SOAP fault element as the body entry, then an error has occurred. In this case, the accuracy of any other fields in the response cannot be guaranteed, and you should only use the fault sub-elements to identify the error. Currently, these sub-elements are as follows:

• **faultcode** is intended for use by the client software and provides an algorithmic mechanism for identifying a fault.



• **faultstring** provides a human-readable explanation of a fault, and is not intended for algorithmic processing.

JSON API

As an alternative to XML API, PortaSwitch® supports JSON API, thus providing your development department with a choice of Web Application Services that can be used. Among the advantages of JSON API are the following:

- Simple data structures that can be easily read and written.
- JSON format is faster in parsing and generating data due to simple syntax, thus there is little influence on web server performance.
- Supports the same methods as those in the SOAP.
- Simplifies the creation of front-end web sites that receive and modify data with minimum impact on performance.

Access to JSON API

All JSON requests to PortaSIP® Media Server API must be sent to the following URL: https://<web-

server.yourdomain.com>:<port>/rest/<service>/<method>/



Please replace the **web-server.yourdomain.com** with the actual hostname of your PortaSwitch® web server.

Replace <port> with the required port. The JSON interface is available for administrators on port 443, the interface for customers is available on port 8444, the interface for resellers is available on port 8442 and the interface for accounts is available on port 8445.

Replace **<service>** with the API service that contains the required method (e.g. specify the **SMPreferences** service to manage voice mailbox preferences.)

Replace <method> with the required API method (e.g. specify set_folder_preferences method in order to change mailbox folder preferences.)

Here is an example of the URL the POST request to be sent to:

https://demo.portaone.com:8443/rest/SMPreferences/set_folder_preferences/



Sending an HTTP request

For HTTP requests you must include the following parameters (in JSON format) in the POST request body:

- auth_info The mandatory authentication. Not used with the methods to establish the API session information (see the Security section).
- params A set of method parameters (in JSON format) that depend on a method structure. Note that method parameters and their structures are the same as those in the SOAP.

The Content-Type header field used with a HTTP POST request must have one of the following values:

- application/x-www-form-urlencoded
- multipart/form-data

Please note that special characters must be escaped. For example, if you want to send the HTTP GET request:

```
https://111.111.11.11:8443/rest/AutoAttendant/set_menu_transiti
on/{"login":"000111222","password":"mysEcReTp@ss","session_id":
null,"domain":"111.111.11.11"}/{"i_menu":12,"transition_info":"
event":"#","action":"Transfer","target_i_menu":0,"destination":
"1","play_prompt":"Y"}}
```

it must be transformed into the following form:

https://111.111.11.11:8443/rest/AutoAttendant/set_menu_transition/%7B%22login%22%3A%22000111222%22,%22password%22%3A%22mysEcReTp@ss%22,%22session_id%22%3Anull,%22domain%22%3A%22111.11.11.11%22%7D/%7B%22i_menu%22%3A12%22transition_info%22%3A%7B%22event%22%3A%22%2523%22%2C%22action%22%3A%22Transfer%22%2C%22target_i_menu%22%3A0%2C%22destination%22%3A%221%22%2C%22play_prompt%22%3A%22Y%22%7D%7D

You can run JSON requests in the dry run mode. The dry run mode does not execute the method itself. Instead, it checks input arguments according to the schema validation rules and returns validation results. To run a request in the dry run mode, add the aux_info structure into the request. The structure has the following fields:

• dry_run – indicates that the method must be run in the dry run mode.

Examples of API requests

The examples below are given with the use of cURL command line tool.

establish API session

Request:

```
curl https://demo.portaone.com:8444/UM/rest/Session/login
-d auth_info='{}'
-d params='{"login":"SIPAccounts", "password":"123password"}'
```



Response:

{"session id":"flab18fe5a3decf0ba828e56a3d9e982"}

Error handling

If the server returns the '500 Internal Server Error' status code in the HTTP response, then the response body contains a JSON object which includes two elements (keys) that carry error information:

- **faultcode**, that is intended for use by the client software and provides an algorithmic mechanism for identifying the fault.
- **faultstring**, that provides a human readable explanation of the fault, and is not intended for algorithmic processing.

WSDL

Each PortaSIP® Media Server has its own set of WSDL documents available for download from the web server. These documents can be downloaded from:

- https://web
 - server.yourdomain.com:8443/soap/wsdl.fcgi?get=Session.xsd
- https://web
 - server.yourdomain.com:8443/soap/wsdl.fcgi?get=Types.xsd
- https://web
 - server.yourdomain.com:8443/soap/wsdl.fcgi?get=Voicemail.xsd
- https://web
 - server.yourdomain.com:8443/soap/wsdl.fcgi?get=SMPreferences.xsd
- https://web
 - server.yourdomain.com:8443/soap/wsdl.fcgi?get=DialDirectory.xsd
- https://web
 - server.yourdomain.com:8443/soap/wsdl.fcgi?get=AutoAttendant.xsd
- https://web
 - server.yourdomain.com:8443/soap/wsdl.fcgi?get=Conference.xsd

All requests to PortaSIP® Media Server API are handled via an SSL connection. By default, PortaSIP® Media Server installations contain a self-signed certificate that provides the means to encrypt data. However, since this certificate's authenticity cannot be validated, you may experience some problems when connecting to an SSL site. In that case, it may be necessary to obtain a certificate from a genuine certificate authority. Another option is to generate your own certificate authority and have certificates deployed to all API clients. However, this goes beyond the scope of the present document.



2. Reference



Notation conventions

The following typographic conventions apply throughout this chapter:

- * A value can be entered for this property only when inserting new records and cannot be changed later.
- ** This property is read-only, and its value cannot be changed.
- Mandatory properties (whose value must be entered during insertion, and cannot be set to an empty value later) are underlined.
- n This property can be used with the **nil** attribute to indicate that it is blank (has no content):
 - In the Request message the xsi:nil="true" attribute can be used to clear the property (set value to NULL in the database).
 - In the Response message a property has the xsi:nil="true" attribute if it is blank (has the NULL value in the database).

Establishing an authenticated session

SOAP URI: https://web-server.yourdomain.com/UM/SOAP/Session

Methods

login

Parameters: LoginRequest Return value: LoginResponse

Checks the validity of login and password and returns session_id on success. An API fault is generated on failure.

logout

Parameters: LogoutRequest Return value: LogoutResponse

Terminates the session. You should call logout() to terminate the session properly.



Type reference

LoginRequest structure

Property	Type	Description
login	string, 32 char	Account ID specified on web
	max	interface
domain	string	Media Server Domain
		corresponding to billing
		environment that the account
		belongs to
password	string, 16 chars	Password specified on web
	max	interface

LoginResponse structure

Property	Type	Description
session_id	string, 32 chars	ID of newly opened session
	max	

LogoutRequest structure

Property	Type	Description
-	-	-

LogoutResponse structure

Property	Type	Description
success	int	1 in case of success, 0 in case of
		failure

Global methods and types

Type reference

The structure below is used to pass authentication data to the API method. There are two possible ways to authenticate an API request: create a session and pass session_id in auth_info, or pass all the required credentials (login/domain/password) in every API request.

auth_info structure

Property	Type	Description
login	string, 32 chars max	Account ID specified on web
		interface



domain	string	Media Server Domain
		corresponding to current billing
		environment
password	string, 16 chars max	Account's password for web self-
		care interface
or alternatively:		
session_id	string, 32 chars max	The unique ID of previously
		opened API session

Voicemail Settings

SOAP URI: https://web-server.yourdomain.com/UM/SOAP/Voicemail

Methods

get_vm_settings

Parameters: GetVMSettingsRequest Return value: GetVMSettingsResponse

This method enables an API user (account) to get general voicemail settings from the PortaSIP® Media Server database.

set_vm_settings

Parameters: SetVMSettingsRequest Return value: SetVMSettingsResponse

This method enables an API user (account) to set general voicemail settings in the PortaSIP® Media Server database.

get_vm_greeting

Parameters: GetVMGreetingRequest Return value: GetVMGreetingResponse

This method enables an API user (account) to get the sound prompt for a specified greeting from the PortaSIP® Media Server database. The sound file is returned in a MIME attachment.

set_vm_greeting

Parameters: **SetVMGreetingRequest**Return value: **SetVMGreetingResponse**



This method enables an API user (account) to set the sound prompt for a specified greeting type. The sound file is sent in a MIME attachment.

Type reference

GetVMSettingsRequest structure

This method doesn't have any parameters.

GetVMSettingsResponse structure

Property	Type	Description
vm_settings	VMSettings	Complete information about
		general voicemail settings

SetVMSettingsRequest structure

May include **any** of the following properties:

Property	Type	Description
vm_settings	VMSettings	Complete information about
		general voicemail settings

SetVMSettingsResponse structure

Property	Type	Description
vm_settings_saved	int	1 in case of success

VMSettings structure

Property	Type	Description
password	string	Password for accessing
		voicemail via IVR
password_ask	string	• yes – ask for password
		when accessing
		voicemail via IVR;
		• no – don't ask for
		password when
		accessing voicemail via
		IVR
prompt_levels	string	PortaSIP® Media Server offers
		three voice prompt settings in
		each supported language:
		• standard
		 extended
		• rapid
announce_dt	string	Announce the date and time
		when each voicemail was sent.



		Values:
		• yes
		• no
auto_play	string	Auto-play new voicemail(s) when a call to voicemail is established. Values: • yes
anatinaa	atrina	Type of amorting for years
greetings	string	Type of greeting for users wishing to leave a voicemail. Values: • standard • extended • personal; • name
fax_file	string	Format for received faxes:
		multi_pngmulti_tiffpdftiff
ext_email	string, max 128 chars	External email for forwarding, copying, and notifying
ext_email_action	string	Action for external email:
ext_email_vm_fmt	string	Voice message audio format: • au • mp3 (default) • wav
enable_disa	string (Yes/No)	Enable DISA functionality for customer's voicemail
disa_password	string	Password for using DISA functionality

GetVMGreetingRequest structure

Property	Type	Description
greeting_type	string	Values:
		• standard
		 extended
		 personal



GetVMGreetingResponse structure

Property	Type	Description
filename	string	Filename of greeting attached to
		SOAP response in a MIME
		attachment

SetVMGreetingRequest structure

Property	Type	Description
greeting info	GreetingInfo	Complete information about
	structure	general greeting's settings

GreetingInfo structure

Property	Type	Description
greeting type	string	Values:
		 extended
		 personal
		• name
<u>filename</u>	string	Filename of greeting attached to
		SOAP request in a MIME
		attachment

SetVMGreetingResponse structure

Property	Type	Description
success	int	1 in case of success
i_audio_file	int	The ID of the audio file in the
		CodecConverter conversion
		queue

Folder preferences and Mailbox and message display options

SOAP URI: https://web-server.yourdomain.com/UM/SOAP/SMPreferences

Methods

get_folder_preferences

Parameters: GetFolderPreferencesRequest Return value: GetFolderPreferencesResponse



This method enables an API user (account) to get the preferences of his mailbox.

set_folder_preferences

Parameters: SetFolderPreferencesRequest Return value: SetFolderPreferencesResponse

This method enables an API user (account) to set the preferences of his mailbox.

get_display_preferences

Parameters: GetDisplayPreferencesRequest Return value: GetDisplayPreferencesResponse

This method enables an API user (account) to get the display preferences of his mailbox and messages.

set_display_preferences

Parameters: SetDisplayPreferencesRequest Return value: SetDisplayPreferencesResponse

This method enables an API user (account) to set the display preferences of his mailbox and messages.

Type reference

GetFolderPreferencesResponse structure

Property	Type	Description
folder_prefs	FolderPreferences	Complete information about the
	structure	folder preferences; for more
		information, see below

FolderPreferences structure

Property	Type	Description
trash_folder	string	An IMAP folder where messages
		are moved on deletion. The messages are
		deleted completely if this field is set
		to "none"
draft_folder	string	An IMAP folder where the user can save
		a message in progress as a draft. The
		messages aren't saved if this field is set
		to "none"
unseen_type	int	The Unread Message Notification Type:
		• 1 – Only Unseen



		• 2 – Unseen and Total
unseen_notify	int	Enable Unread Message Notification:
		• 1 – No Notification
		• 2 – Only INBOX
		• 3 – All Folders
sent_folder	string	An IMAP folder messages are copied to
		after they are sent. The messages aren't
		copied if this field is set to "none"
unseen_cumulative	int	Enable the Cumulative Unread Message
		Notification. This controls the behavior
		of the message counter displayed next to
		each folder in the folder list. When
		enabled, if the folder contains sub-folders
		and is collapsed, then the message count
		includes all messages within all the sub-
		folders of that folder.
search_memory	int	Memory Search options. If the user
		searches the mailbox, the search
		can be saved for quick access later on.
		This option defines how many mailbox
		searches will be saved.
		Allowed Values: 0 (disabled), 1, 2, 3, 4, 5, 6, 7, 8, 9

SetFolderPreferencesRequest structure

Property	Type	Description
folder prefs	FolderPreferences	Complete information about the
	structure	folder preferences; for more
		information, see below

SetFolderPreferencesResponse structure

Property	Type	Description
success	int	1 in case of success

GetDisplayPreferencesRequest structure

GetDisplayPreferencesResponse structure

Property	Type	Description
display prefs	DisplayPreferences	Complete information about
	structure	the display preferences; for
		more information, see below



DisplayPreferences structure

Property	Type	Description
wrap_at	int	Defines how many characters to allow
		before wrapping text
truncate_sender	int	Specifies the lenght of the From / To fields (0 for full)
show_xmailer_default	int	When viewing a message, this displays which email service or client the sender used
editor_height	int	Specifies the height of the Editor Window
mdn_user_support	int	Specifies whether to enable the Mail Delivery Notification
truncate_subject	int	Specifies the lenght of the Subject Field (0 for full)
body_quote	string	Prefix each line of the original message with this symbol when replying or forwarding an email message
include_self_reply_all	int	Specifies whether to include user's address in CC when he chooses Reply All
sig_first	int	Specifies whether to prepend signature before Reply/Forward text
pf_cleandisplay	int	Specifies whether to display the View Printable Version link in a message
editor_size	int	Specifies the width of the Editor Window.
show_html_default	int	Specifies what version to show by default if a received message is sent in both text and HTML formats: 0 – Text version 1 – HTML version
page_selector_max	int	Specifies the maximum number of pages that will be shown at one time
internal_date_sort	int	Specifies whether to sort messages by Received Date
page_selector	int	Specifies whether to show Page Selector. When enabled, message pages will be shown above and below the list of messages, allowing the user to quickly jump to a specific message page
addrsrch_fullname	string	Specifies the format of addresses added from the address book: • "Noprefix" – No prefix, address only



		 "Nickname" – Nickname and address "Fullaname" – Full name and address
show_num	int	Specifies the number of messages that will be shown on one page
show_images	int	Specifies whether to display attached images with the message

SetDisplayPreferencesRequest structure

Property	Type	Description
display_prefs	DisplayPreferences	Complete information about
	structure	the display preferences; for
		more information, see below

SetDisplayPreferencesResponse structure

Property	Type	Description
success	int	1 in case of success

Auto attendant configuration

SOAP URI: https://web-server.yourdomain.com/UM/SOAP/AutoAttendant

Methods

get_menu_list

Parameters: **GetMenuListRequest** Return value: **GetMenuListResponse**

This method enables an API user (account) to get a list of all configured auto attendant menus.

update_menu

Parameters: UpdateMenuRequest Return value: UpdateMenuResponse

This method enables an API user (account) to update the settings of a separate auto attendant menu.

create_menu

Parameters: CreateMenuRequest



Return value: CreateMenuResponse

This method enables an API user (account) to create an auto attendant menu.

del_menu

Parameters: DelMenuRequest Return value: DelMenuResponse

This method enables an API user (account) to delete an auto attendant menu.

del_menu_transition

Parameters: DelMenuTransitionRequest Return value: DelMenuTransitionResponse

This method enables an API user to delete an auto attendant menu transition.

set_menu_prompt

Parameters: SetMenuPromptRequest Return value: SetMenuPromptResponse

This method enables an API user (account) to set (record) separate promt for selected auto attendant menu. The sound file is sent in a MIME attachment to the API request.

get_menu_prompt

Parameters: GetMenuPromptRequest Return value: GetMenuPromptResponse

This method enables an API user (account) to get a separate prompt from the selected auto attendant menu. The sound file is sent in a MIME attachment to the API request.

get_menu_transition_list

Parameters: GetMenuTransitionListRequest Return value: GetMenuTransitionListResponse

This method enables an API user (account) to get a list of auto attendant menu transitions.

set_menu_transition

Parameters: SetMenuTransitionRequest



Return value: SetMenuTransitionResponse

This method enables an API user (account) to set auto attendant menu transitions. The transition prompt should be sent in a MIME attachment.

get_menu_transition_prompt

Parameters: GetMenuTransitionPromptRequest Return value: GetMenuTransitionPromptResponse

This method enables an API user (account) to get an auto attendant menu transition prompt. The prompt is sent in a MIME attachment.

set_menu_transition_promt

Parameters: SetMenuTransitionPromptRequest Return value: SetMenuTransitionPromptResponse

This method enables an API user to set an auto attendant menu transition prompt. The transition prompt should be sent in a MIME attachment.

Type reference

GetMenuListRequest structure

Property	Type	Description
-	-	-

GetMenuListResponse structure

Property	Type	Description
menu_list	array of	The list of auto attendant menus
	MenuInfo	
	structures	

UpdateMenuRequest structure

Property	Type	Description
menu_info	MenuInfo	Auto attendant menu data

UpdateMenuResponse structure

Property	Type	Description
i_menu	int	The unique ID of updated menu record

CreateMenuRequest structure

Property	Type	Description
menu_info	MenuInfo	Auto attendant menu data



CreateMenuResponse structure

Property	Type	Description
i_menu	int	The unique ID of created menu record

DelMenuRequest structure

Property	Type	Description
i_menu	int	The unique ID of deleted menu record

DelMenuResponse structure

Property	Type	Description
i_menu	int	The unique ID of deleted menu database record

DelMenuTransitionRequest structure

Property	Type	Description
i_menu_transition	int	The unique ID of the menu transition
		record

DelMenuTransitionResponse structure

Property	Type	Description
i_menu_transition	int	The unique ID of the menu transition
		record

MenuInfo structure

Property	Type	Description
i_menu*	int	The unique ID of menu record
		(required for the update_menu
		and del_menu methods)
name	string, max 64	The unique within one
	chars	account menu name; 'ROOT'
		name is reserved for the root
		menu, which always exists
period	string, max 255	Period in special format (see
	chars	the How to Define a Time Period
		section of this guide).
period_desc	string, max 255	Description of period in a form
	chars	understandable by end-users
msg_disabled_type	string	'Unavailable' prompt type –
		standard or recorded by user.
		Values:
		• standard
		• custom
msg_timeout_type	string	'Timeout' prompt type –



		atandand on magaint-thi
		standard or recorded by user. Values:
		• standard
		• custom
msg_intro_set	int	1 if 'Into' prompt recorded;
		otherwise 0
msg_menu_set	int	1 if 'Menu' prompt recorded;
		otherwise 0
msg_disabled_set	int	1 if 'Unavailable' prompt
		recorded; otherwise 0
msg_timeout_set	int	1 if 'Timeout' prompt
		recorded;
		otherwise 0
msg_intro_type	string	'Intro' prompt type – standard
		or recorded by user.
		Values:
		• standard
		• custom
msg_menu_type	string	'Menu' prompt type – standard
		or recorded by user.
		Values:
		• standard
		• custom
replay_menu_times	int	The number of times to repaly
replay_mena_emes		the menu prompts
first_digit_timeout	int	The timeout in seconds to wait
n		while the first digit is entered
next_digit_timeout	int	The maximum timeout in
next_digit_timeout	IIIC	seconds between collected
		digits. Default: 5
direct_dial_enabled	string (Y/N)	If set to Y, allow dialing
	(-/-1)	extension from the menu
		directly. If enabled, the
		"DirectDial" value for the
		action attribute will be
		forbidden.
		Default: N

SetMenuPromptRequest structure

Property	Type	Description	
i_menu	int	The unique ID of updated menu record	
prompt_type	string	Prompt type:	
		• intro	
		• menu	



		disabled
		• timeout
prompt	string	Filename of a prompt that is being sent in a
		MIME attachment to the API request

SetMenuPromptResponse structure

Property	Type	Description
i_menu	int	The unique ID of updated menu record
i_audio_file	int	The ID of the audio file in the
		CodecConverter conversion queue

GetMenuPromptRequest structure

Property	Type	Description
i_menu	int	The unique ID of menu record
prompt_type	string	Prompt type:
		• intro
		• menu
		disabled
		• timeout

GetMenuPromptResponse structure

Property	Type	Description
prompt	string	Filename of a prompt that is being sent in a MIME attachment to the API response

GetMenuTransitionListRequest structure

Property	Type	Description
<u>i menu</u>	int	The unique ID of menu record

GetMenuTransitionListResponse structure

Property	Type	Description
transition_list	array of	Set of transitions for specified
	TransitionInfo	auto attendant menu
	structures	

SetMenuTransitionRequest structure

Property	Type	Description
<u>i menu</u>	int	The unique ID of the menu
		record
transition info	TransitionInfo	Properties of the menu transition



SetMenuTransitionResponse structure

Property	Type	Description
<u>i_menu_transition</u>	int	The unique ID of the menu
		transition record
i_audio_file	int	The ID of the audio file in the
		CodecConverter conversion
		queue

GetMenuTransitionPromptRequest structure

Property	Type	Description
<u>event</u>	string	Transition event; see allowed
		values in TransitionInfo
		structure
<u>i menu</u>	int	The unique ID of menu record
i_menu_transition	int	The unique ID of the menu
		transition record

GetMenuTransitionPromptResponse structure

Property	Type	Description
prompt	string	Filename of a prompt that is
		being sent in a MIME attachment
		to the API request

SetMenuTransitionPromptRequest structure

Property	Type	Description
i menu transition	int	The unique ID of the menu
		transition record
<u>prompt</u>	string	Filename of a prompt that is
		being sent in a MIME
		attachment to the API request

SetMenuTransitionPromptResponse structure

Property	Type	Description
i menu transition	int	The unique ID of the menu
		transition record
i_audio_file	int	The ID of the audio file in the
		CodecConverter conversion
		queue

TransitionInfo structure

Property	Type	Description
action	string	Performed action.



		Possible values:
		Disabled – No action.
		Directory – Launch
		the 'Dial Directory'
		IVR.
		• Queue – Transfer to
		the call queue
		specified in the
		target_i_menu property.
		• Transfer – Transfer to
		the preconfigured
		number specified in
		the destination
		property.
		• TransferE164 –
		Transfer to the E164
		number specified in
		the destination
		property.
		Voicemail – Launch voicemail – Launch
		voicemail recording.
		Menu – Go to the
		auto attendant menu
		specified in
		target_i_menu property.Extension – Transfer
		to the extension
		dialed by a user. Note
		that at the voice
		prompt request, the
		user must input a
		menu item first and
		then the extension
		number.
		• DISA – Make a call.
		DirectDial – Transfer
		to the extension
		dialed by a user. Note
		that the first number
		of the extension must
		coincide with the
		current action digit.
		DisconnectCall –
		Disconnect a call.
announce_ext_numbers	string	Specifies whether to
		announce the external



		number.
		Possible values: • Y – Announce the external number. • N – Don not announce the external number.
destination	string, max. 32 chars	Destination for 'Transfer,' 'TransferE164' action
event	string	Transition event.
		Possible values: '0', '1', '2', '3', '4', '5', '6', '7', '8', '9', '*', '#', 'Timeout', 'Not Active', 'f'.
i_menu_transition	int	The unique ID of the menu transition record
max_size	int	The maximum allowed number of digits that a user can input as an extension (applicable only for the Extension <i>action</i>)
play_prompt	string	Play or do not play user-recorded prompt before action. Possible values: Y N
prompt	string	Filename of a user-recorded prompt that is being sent in a MIME attachment (only for the set_menu_transition method)
prompt_set	int	1 if user-recorded prompt set
target_i_menu	int	The unique ID of the auto attendant menu record
target_i_queue	int	The unique ID of the call queue to which the call must be transferred.



Conference configuration

SOAP URI: https://web-server.yourdomain.com/UM/SOAP/Conference

Methods

get_conf_info

Parameters: GetConfInfoRequest Return value: GetConfInfoResponse

Realm: account

This method enables an API user to obtain conference settings by i_conf or name.

get_conf_list

Parameters: GetConfListRequest Return value: GetConfListResponse

Realm: account

This method enables an API user to obtain a list of all his conferences and their settings.

create_conf

Parameters: CreateConfRequest Return value: CreateConfResponse

Realm: account

This method enables an API user to create a new conference entity.

update_conf

Parameters: UpdateConfRequest Return value: UpdateConfResponse

Realm: account

This method enables an API user (account) to update a conference entity.

del_conf

Parameters: DelConfRequest Return value: DelConfResponse

Realm: account



This method enables an API user to delete a certain conference.

set_conf_prompt

Parameters: SetConfPromptRequest Return value: SetConfPromptResponse

This method enables an API user (account) to set (record) separate prompts for conferences. The sound file is sent in a MIME attachment to the API request.

get_conf_prompt

Parameters: GetConfPromptRequest Return value: GetConfPromptResponse

This method enables an API user (account) to get a prompt recorded for a conference. The sound file is sent in a MIME attachment to the API request.

Type reference

Confinfo structure

Property	Type	Description
i_conf	int	The unique ID for a conference
		entity
name	string	A conference name
pin_host	string	PIN for administrator to log into
		the conference
pin_user	string	PIN for user to join the
		conference
max_call_duration	int	Maximum conference duration
max_session_time	int	Maximum session time (exclusive
		with start / end time)
max_participants	int	Maximal participants
wait_host	string	Whether the administrator should
		log in first. Allowed values: Y, N
play_announce	string	Specifies whether announcements
		should be played.
		Allowed values: Y, N
play_moh	string	Specifies whether MOH should
		be played. Allowed values: Y, N
start_time	string	Specifies when the conference
		will start (For permanent
		conference use
		max_session_time). Note that the
		time is defined in UTC



msg_intro_set	string	1 if an 'Intro' prompt recorded; otherwise 0
moh_set	string	1 if a 'MOH' prompt recorded;
		otherwise 0
video_conf	string	Specifies whether video
		conference is enabled. Allowed
		values: Y, N

GetConfInfoRequest structure

Property	Type	Description
i_conf	int	The unique ID for the conference
name	string	The conference name

GetConfInfoResponse structure

Property	Type	Description
conf_info	ConfInfo structure	General conference settings

GetConfListRequest structure

Property	Type	Description
-	-	-

GetConfListResponse tructure

Property	Type	Description
conf_list	array of	The list of conferences and their
	ConfInfo	settings
	structures	

CreateConfRequest structure

Property	Type	Description
conf_info	ConfInfo structure	General conference settings

CreateConfResponse structure

Property	Type	Description
i_conf	int	The unique ID for a new
		conference

UpdateConfRequest structure

Property	Type	Description
conf_info	ConfInfo structure	General conference settings



UpdateConfResponse structure

Property	Type	Description
i_conf	int	The unique ID for the updated
		conference

DelConfRequest structure

Property	Type	Description
i_conf	int	The unique ID for the conference
		to be deleted

DelConfResponse structure

Property	Type	Description
i_conf	int	The unique ID for deleted
		conference

SetConfPromptRequest structure

Property	Type	Description
<u>i_conf</u>	int	The unique ID for a conference
		record
prompt_type	string	Prompt type:
		• intro
		• moh
prompt	string	Filename for a prompt that is
		being sent in a MIME attachment
		to the API request

SetConfPromptResponse structure

Property	Type	Description
<u>i conf</u>	int	The unique ID for the updated
		conference record
i_audio_file	int	The ID of the audio file in the
		CodecConverter conversion
		queue

GetConfPromptRequest structure

Property	Type	Description
<u>i conf</u>	int	The unique ID for a conference
		record
prompt_type	string	Prompt type:
		• intro
		• moh



GetConfPromptResponse structure

Property	Type	Description
prompt	string	Filename of a prompt that is
		being sent in a MIME attachment
		to the API response



3. Call control API



Overview

The Call control API permits to begin, answer and terminate a call, retrieve a list of currently established calls and subscribe to notifications about call state changes for the whole IP Centrex environment as well as for individual extensions. Together with already existing API methods (e.g. to retrieve customer information), these help to build a full-grown CTI solution.

The Call control API is accessible via WebSockets. WebSocket connections are processed by workers. Each worker can process up to 100 concurrent connections. The actual maximum number of connections possible, however, depends on the capacity and general configuration of the Apache server.

Access to JSON-RPC API

All JSON-RPC requests to the API must be sent to the following URL: wss://<web-server.yourdomain.com>:<port>/ws



Please replace the **web-server.yourdomain.com** with the actual hostname of your web server.

Replace **port**> with the required port. The JSON-RPC interface is available for administrators on port 443, the interface for customers is available on port 8444, the interface for resellers is available on port 8442 and the interface for accounts is available on port 8445.

Here is an example of the URL the POST request to be sent to:

Sending a JSON-RPC request

For JSON-RPC requests you must include the following parameters in the POST request body:

- cseq Since the WebSocket protocol is asynchronous, this value is used to match the response with the request (the same value is present in the response). If no value is passed in the request, no response is expected and none will be returned.
- auth_info The mandatory authentication information (see the Security section).
- service The API service that contains the required method.
- method The name of the required API method.
- params A set of method parameters (in JSON format) that depend on a method structure.



The example below illustrates the login request:

The response contains the session ID value:

Error handling

In case a request could not be executed or had errors in its structure, the response contains the following error information:

- **code**, that is intended for use by the client software and provides an algorithmic mechanism for identifying the fault.
- **message**, that provides a human readable explanation of the fault, and is not intended for algorithmic processing.
- **details**, that complement the **message** and contain the erroneous object.

Error codes are listed in the table below:

Service	Error code	Error	Details
		message	
CallControl	sip.unsupported_method	Unsupported	The specified
		method	method name
			is unknown
CallControl	sip.wrong_parameters	Bad	Incorrect
		parameters	parameter list
			- required
			parameters are
			missing or
			contain
			incorrect
			values
CallControl	sip.internal_server_error	Internal	Unspecified
		processing	processing



		error	error prevents
			the correct
			execution of a
			method
CallControl	sip.call_not_found	Not found	A call (call
			part) specified
			by session_id
			(dialog_id)
			was not found
CallControl	sip.call_control_disabled	Disabled	The used
	_		method is
			disabled
CallControl	sip.service not enabled	The service	The
	-	is not	conferencing
		enabled	service is not
			enabled for
			this account
all	internal_error	Internal	
		server error	

Call state notification management

URL (namespace): wss://portabillingweb.yourdomain.com/ws/CallControl

These methods enable an agent to monitor calls in progress (outgoing and incoming) and receive notifications about call state changes. This helps in manipulating calls (e.g. redirect the call to another party if the extension is busy).

Use these methods together with the **Voice API** methods to build your CTI solutions such as attendant console or click-to-dial application for your CRM system.

The call states can be monitored for:

- an individual extension. It is represented as an account in PortaSwitch® and is identified by the i account key;
- an entire IP Centrex environment. It is represented as a customer in PortaSwitch® and is identified by the i customer key;
- an access number of your custom IVR application. It is identified by the i_ivr_an key.
- an external number. Subscribe the app (e.g. a switchboard app) to call state notifications for the account within your IP Centrex in the usual way. When an account transfers, forwards or makes a



call, the app receives call state notifications from external numbers via that account's channel.

To retrieve the internal ID value of either entity, use **PortaBilling API**. For example, to find i_account of an office extension you wish to receive notifications for, call the get_account_list API method:

```
"cseq": 2,
    "service": "Account",
    "method": "get_account_list",
    "auth_info":{
        "session_id":"b11c226be16aa179b0d6b2fa0fd1394c"
        },
        "params": {
        "id":"12065558954"
    }
}
```

You can find the information about required PortaBilling® API methods in the descriptions to the method attributes.

Methods

enable_api_notifications

This method enables an agent to subscribe and receive call state notifications for:

- an individual extension by passing the i_account value in the API request;
- the entire IP Centrex environment by passing the i_customer value in the API request;
- the main or a branch office within the IP Centrex environment. This applies when a company has independent offices (i.e. branches) linked to the main one. Such offices are identified by the i_main_office value.
- the access number of your custom IVR application by passing the i_ivr_an value in the API request.

When a call state changes, the sip.call_control_notifications event is sent. It contains the current call state.

Parameters: EnableApiNotificationsRequest Return value: EnableApiNotificationsResponse Realm: administrator, reseller, retail customer, account

Request example:



```
"cseq": 2,
   "auth_info": {
        "session_id": "f9d7eab82631b385fbecff9b65883076"
},
   "service": "CallControl",
   "method": "enable_api_notifications",
   "params": {
        "event": "sip.call_control_notifications",
        "i_customer": 30
}
```

Response example:

```
{
    "cseq":2,
    "result":{
        "success":1
    },
    "success":1
}
```

Example of a sip.call_control_notifications event



Example of a sip.call_control_notifications event for auto attendant account.

```
"connect time": "2018-12-28 13:12:25.245",
           "dialog id": [
                 "7e033351-4e65ab85@192.168.64.196",
                 "112347"
          ],
"direction":
"recipient",
           "local party": {
                "account_id": "123010",
"centrex_id": "30",
                "display id": "123010",
                "forwarder list":[
     "account id": "123010",
      "centrex id": "30",
      "display id": "123010",
      "forwarders": [],
      "id": "123010"
                "id": "123010",
"net": "0"
                "operators": 2,
                "position": 0
          "remote_party": {
    "account_id": "11198700001",
    "centrex_id": "10",
    "display_id": "11198700001",
                "forwarders": [],
                "id": "11198700001",
                "name": "11198700001",
                 "net": "0"
          "session_id": 192123,

"sip_transport_id": "192.168.67.30:5070",

"start_time": "2018-12-28 13:12:25.146",

"state": "queued"
```

Example of a sip.call_control_notifications event for an account that forwards a call to an external number.

```
{
  "action":"update",
  "event":"sip.call_control_notifications",
  "result":{
  "call_info":{
  "call":{
  "id":"b0fc4957-6fc3a86f@192.168.233.134",
  "tag":"qdef43kz9zxym4lz.o"
  },
  "callee":{
  "account_id":"123007",
```



```
centrex id":"30",
"display id":"123007",
"forwarder list":[
    "account id": "123008",
    "centrex id": "30",
     "display id": "123008",
     "id": "123008"
"centrex id": "30",
"display_id": "123009",
"forwarders": [],
    "id": "123009"
"id":"123007"
"account id":"123002",
"display_id":"123002",
"forwarder list":[
"reason": "Call terminated by API",
"state":"terminated",
"type": "incoming"
```

disable api notifications

This method enables an agent to unsubscribe from call state notifications for:

- an individual extension by passing the i_account value in the API request;
- the entire IP Centrex environment by passing the i_customer value in the API request;
- the main or a branch office within the IP Centrex environment. This applies when a company has independent offices (i.e. branches) linked to the main one. Such offices are identified by the i_main_office value.
- the access number of an IVR application by passing the i_ivr_an value in the API request.

Parameters: **DisableApiNotificationsRequest**Return value: **DisableApiNotificationsResponse**Realm: administrator, reseller, retail customer, account



Request example:

```
"cseq": 2,
    "auth_info": {
        "session_id": "f9d7eab82631b385fbecff9b65883076"
    },
    "service": "CallControl",
    "method": "disable_api_notifications",
    "params": {
        "event": "sip.call_control_notifications",
        "i_customer": 30
    }
}
```

Response example:

```
{
   "cseq":2,
   "result":{
        "success":1
   },
   "success":1
}
```

Type reference

EnableApiNotificationsRequest structure

The request must contain at least one attribute that is mentioned in the structure below.

Property	Type	Description
i_account	unsignedLong	The unique ID of the account record.
		The account represents a phone line or
		an office extension.
		To get the account ID, call the
		get_account_list method. The
		<i>i_account</i> is returned in the
		AccountInfo structure.
		See
		https://www.portaone.com/docs/Porta
		Billing_API.html#AccountInfo.
i_ivr_an	unsignedLong	The unique ID of the access number
		associated with the custom IVR
		application (it has the User application
		type on the PortaBilling® web
		interface).
		In DouteDilling® the IVD agence
		In PortaBilling® the IVR access



		number is also associated with the
		account record. This is required to
		apply charges for using this number.
		Call the obtain_access_number API
		method to assign the access number to
		the account and receive its ID in the
		response.
		See
		https://www.portaone.com/docs/Porta
		Billing_API.html and
		https://www.portaone.com/docs/Porta Billing_API.html#ObtainAccessNumber
		Response.
		Response.
		If you operate under the customer,
		reseller or the administrator realm, first
		retrieve the <i>i_account</i> to which you wish
		to assign the access number and pass it
		in the API request.
i_customer	unsignedLong	The unique ID of the customer record.
		To get the customer ID, call the
		get_customer_info method. The
		<i>i_customer</i> is returned in the
		CustomerInfo structure.
		See
		https://www.portaone.com/docs/Porta
		Billing_API.html#CustomerInfo.
i_main_office	unsignedLong	The unique ID of the main office
		(customer record with office type 3).
		To get the main office ID and
		headquarters office type, call the
		get_customer_info method.
		Possible values for the <i>i_office_type</i>
		attribute in the response are the following:
		• 1 – none
		• 2 – branch office
		• 3 – main office
		If the office type is 1 (none), leave this
		attribute empty.
		If the office type is 2 (branch office),
		specify the office ID from the



i_main_office attribute.
If the office type is 3 (main office), specify the main office ID from the <i>i_customer</i> attribute.
See https://www.portaone.com/docs/Porta Billing_API.html#CustomerInfo.

EnableApiNotificationsResponse structure

Property	Type	Description
success	int	A non-zero value means that the
		operation was completed

DisableApiNotificationsRequest structure

The request must contain at least one attribute that is mentioned in the structure below.

Type	Description
unsignedLong	The unique ID of the account record. The account represents a phone line or an office extension. To get the account ID, call the
	get_account_list method. The
	<i>i_account</i> is returned in the AccountInfo structure.
	recountino structure.
	See
	https://www.portaone.com/docs/Porta Billing_API.html#AccountInfo If you
	operate under Customer
unsignedLong	The unique ID of the access number associated with the custom IVR application (it has the User application type on the PortaBilling® web interface).
	In PortaBilling® the IVR access number is also associated with the account record. This is required to apply charges for using this number. Call the obtain_access_number API method to assign the access number to the account and receive its ID in the response.
	unsignedLong



		See https://www.portaone.com/docs/Porta Billing_API.html and https://www.portaone.com/docs/Porta Billing_API.html#ObtainAccessNumber Response.
		If you operate under the customer, reseller or the administrator realm, first retrieve the <i>i_account</i> to which you wish to assign the access number and pass it in the API request.
i_customer	unsignedLong	The unique ID of the customer record. To get the customer ID, call the get_customer_info method. The <i>i_customer</i> is returned in the CustomerInfo structure.
		See https://www.portaone.com/docs/Porta Billing_API.html#CustomerInfo.
i_main_office	unsignedLong	The unique ID of the main office (customer record with office type 3). To get the main office ID and headquarters office type, call the get_customer_info method. Possible values for the <i>i_office_type</i> attribute in the response are the following: • 1 – none • 2 – branch office • 3 – main office If the office type is 1 (none), leave this attribute empty. If the office type is 2 (branch office), specify the office ID from the <i>i_main_office</i> attribute. If the office type is 3 (main office), specify the main office ID from the <i>i_customer</i> attribute. See https://www.portaone.com/docs/Porta



	Billing_API.html#CustomerInfo.
--	--------------------------------

DisableApiNotificationsResponse structure

Property	Type	Description
success	int	A non-zero value means that the
		operation was completed

Voice API

URL (namespace): wss://portabillingweb.yourdomain.com/ws/CallControl

These API methods enable an agent to control call flow so that they can make, receive or redirect calls from their applications or web browsers.

The agent is represented as an account in PortaSwitch® and is charged for making calls.

Methods

get_sip_calls_list

This method enables an agent to receive a list of calls in progress for an individual extension or for the whole IP Centrex environment. For this, subscribe to API notifications using the **enable_api_notifications** method.

Parameters: GetSipCallsListRequest Return value: GetSipCallsListResponse

Realm: administrator, reseller, retail customer, account

Standalone mode support: Yes

Request example:

```
{
    "cseq":2,
    "auth_info":{
        "session_id":"ba6596e4b60919f8695033a20519d6af"
    },
    "service":"CallControl",
    "method":"get_sip_calls_list",
    "params": {
            "i_customer": 30
     }
}
```

Response example:



```
"cseq":2,
           "call":{
              "id": "30108b5e-b29bdab0@192.168.233.134",
              "tag": "ba6783868a4100a8o1"
          },
"callee":{
              "display id":"123007",
              "forwarder list":[
          },
"caller":{
             "account_id":"123002",
"centrex_id":"30",
"display_id":"123002",
"forwarder_list":[
           "start time": "2018-11-29 13:41:04",
           "state": "ringing",
           "type": "outgoing"
           "call":{
              "tag": "qdkr5cjc9cfyzxyb.o"
          },
"caller":{
              "centrex_id":"30",
"display_id":"123002",
              "id":"123002"
           "start time":"2018-11-29 13:41:04",
          "state": "ringing",
          "type":"incoming"
"success":1
```



originate_advanced_call

This method enables an agent to initiate a callback call to a phone number or an extension and then connect it with the desired destination.

The number to which the callback should be established is specified in the caller_id attribute. The destination number is defined in the callee_id attribute. The bill_id attribute contains the ID of the agent's account in PortaSwitch® to charge for this call.

PortaSIP® first places a call to the **caller_id** destination according to the routing plan (leg A). When the first UA answers the call, PortaSIP® places a second call to the destination specified as **callee_id** (leg B).



For calls to go through, the product configuration for the account specified in the **bill_id** attribute must include the rating entry with the INCOMING access code.

Parameters: OriginateAdvancedCallRequest
Return value: OriginateAdvancedCallResponse
Realm: administrator, reseller, retail customer, account

Request example:

```
{
    "cseq":3,
    "auth_info":{
        "session_id":"48e8eedee6a6a520bd5e0380b83ffeb7"
    },
    "service":"CallControl",
    "method":"originate_advanced_call",
    "params":{
        "bill_id":"123007",
        "callee_id":"123003",
        "caller_id":"123007"
    },
}
```

Response example:

```
{
    "cseq":3,
    "result":{
        "call":{
             "id":"BBfc!rA3EV9JNsDxdhRoPvXGW4xR@192.168.243.133"
        },
        "success":1
    },
    "success":1
}
```

Notification example for call leg A:



```
"action": "update",
"event": "sip.call control notifications",
          "id": "BBfc!rA3EV9JNsDxdhRoPvXGW4xR@192.168.243.133
          "tag":"tz+d-zju2fxikkor.o"
       "callee":{
          "account_id":"123007",
"centrex_id":"30",
"display_id":"123007",
"forwarder_list":[
      },
"caller":{
    "cpla
          "display id":"123003",
          "forwarder list":[
       "start_time":"2018-12-03 13:43:01",
"state":"trying",
       "type": "incoming"
"action": "update",
       "call":{
          "id": "BBfc!rA3EV9JNsDxdhRoPvXGW4xR@192.168.243.133
          "tag":"tz+d-zju2fxikkor.o"
      },
"callee":{
          "display id":"123007",
          "forwarder list":[
       "caller":{
          "display_id":"123003",
          "forwarder list":[
       "start time": "2018-12-03 13:43:01",
       "state": "ringing",
```



Notification example for call leg B:



```
"centrex id":"30",
              "display id":"123007",
             "forwarder list":[
          "state": "trying",
          "type": "incoming"
   "action":"update",
   "event": "sip.call control notifications",
             "id": "BBfc!rA3EV9JNsDxdhRoPvXGW4xR@192.168.243.133
             "tag": "tz+d-zju2fw2o4ws.o"
          },
"callee":{
             "account_id":"123003",
"centrex_id":"30",
"display_id":"123003",
"forwarder_list":[
          "display_id":"123007",
          },
"start_time":"2018-12-03 13:43:05",
          "state": "ringing",
          "type": "incoming"
   "action": "update",
          "call":{
             "id": BBfc!rA3EV9JNsDxdhRoPvXGW4xR@192.168.243.133
~10",
             "tag": "tz+d-zju2fw2o4ws.o"
          "callee":{
```



```
"centrex_id":"30",
    "display_id":"123003",
    "forwarder_list":[

    ],
    "id":"123003"
},
    "caller":{
        "account_id":"123007",
        "centrex_id":"30",
        "display_id":"123007",
        "forwarder_list":[

        ],
        "id":"123007"
        },
        "connect_time":"2018-12-03 13:43:08",
        "state":"connected",
        "type":"incoming"
        }
    }
}
```

answer_call

This method enables an agent to answer incoming calls using the application instead of picking up a handset.

Parameters: AnswerCallRequest Return value: AnswerCallResponse

Realm: administrator, reseller, retail customer, account

Request example:

```
{
    "cseq":2,
    "auth_info":{
        "session_id":"9dc0afbab375071f4d132fa82502025c"
},
    "service":"CallControl",
    "method":"answer_call",
    "params":{
        "call":{
            "id":"30108b5e-b29bdab0@192.168.233.134",
            "tag":"qdkr5cjc9cfyzxyb.o"
        },
        "callee_answer_mode":"notify"
}
```

Response example:

```
{
    "cseq":2,
    "result":{
        "success":1
    },
    "success":1
```



Notification example from the party that answers the call

terminate_call

This method enables an agent to disconnect a call from the application.

Parameters: **TerminateCallRequest** Return value: **TerminateCallResponse**

Realm: administrator, reseller, retail customer, account

Request example:

```
{
   "cseq":2,
   "auth_info":{
        "session_id":"9dc0afbab375071f4d132fa82502025c"
   },
   "service":"CallControl",
   "method":"terminate_call",
   "params":{
        "call":{
        "id":"b0fc4957-6fc3a86f@192.168.233.134",
```



```
"tag":"qdef43kz9zxym4lz.o"
},
}
```

Response example:

```
{
    "cseq":2,
    "result":{
        "success":1
    },
    "success":1
}
```

Notification example from the terminated party:

hold_call

This method enables an agent to put a call on hold from the application, without dialing the Hold key on the UA.



The UA must support NOTIFY request with "Event: hold" (see BroadWorks Remote Control Talk Event Package) to initiate hold.

If UA doesn't support the event package, it replies with 400, 489 error code. In this case PortaSIP® places both call parties on hold and plays its own MOH.

Parameters: HoldCallRequest Return value: HoldCallResponse

Realm: administrator, reseller, retail customer, account

Request example:

```
{
   "cseq":2,
   "auth_info":{
        "session_id":"9dc0afbab375071f4d132fa82502025c"
   },
   "service":"CallControl",
   "method":"hold_call",
   "params":{
        "call":{
            "id":"402a8bb-e4e23269@192.168.233.134",
            "tag":"3481849322414871o1"
        },
    }
}
```

Response example:

```
{
    "cseq":2,
    "result":{
        "success":1
    },
    "success":1
}
```

Notification example from the party that places the call on hold:



```
},

"caller":{
        "account_id":"123002",
        "centrex_id":"30",
        "display_id":"123002",
        "forwarder_list":[

        ],
        "id":"123002"
      },
      "state":"holding",
      "type":"outgoing"
    }
}
```

Notification example from the party that is placed on hold:

unhold_call

This method enables an agent to release a call from hold from the application. The UA must support NOTIFY request with "Event: talk" (see BroadWorks Remote Control Talk Event Package) to initiate unhold.



If the UA does not support this event package, it replies with 400, 489 error code. PortaSIP® unholds both call parties and stops playing its own MOH.

Parameters: UnholdCallRequest Return value: UnholdCallResponse

Realm: administrator, reseller, retail customer, account

Request example:

```
{
   "cseq":2,
   "auth_info":{
        "session_id":"9dc0afbab375071f4d132fa82502025c"
   },
   "service":"CallControl",
   "method":"unhold_call",
   "params":{
        "call":{
            "id":"402a8bb-e4e23269@192.168.233.134",
            "tag":"3481849322414871o1"
        },
    }
}
```

Response example:

```
{
    "cseq":2,
    "result":{
        "success":1
    },
    "success":1
}
```

Notification example from the released party:



transfer call

This method enables an agent to perform blind call transfer to another extension or an external number from the application.

Parameters: TransferCallRequest Return value: TransferCallResponse

Realm: administrator, reseller, retail customer, account

Request example:

```
"cseq":2,
    "auth_info":{
        "session_id":"9dc0afbab375071f4d132fa82502025c"
},
    "service":"CallControl",
    "method":"transfer_call",
    "params":{
        "call":{
            "id":"9fb147fc-4f045471@192.168.233.134",
            "tag":"9dwxscwxpshymxri.o"
        },
        "cld": "123003"
}
```

Response example:

```
{
    "cseq":2,
    "result":{
        "success":1
    },
    "success":1
}
```

Notification example for the transferring party to be terminated:

```
{
    "action":"update",
    "event":"sip.call_control_notifications",
    "result":{
        "call_info":{
```



```
"call":{
    "id":"9fb147fc-4f045471@192.168.233.134",
    "tag":"9dwxscwxpshymxri.o"
},
    "callee":{
        "account_id":"123007",
        "centrex_id":"123007",
        "forwarder_list":[

        l,
        "id":"123007"
},
    "caller":{
        "account_id":"123002",
        "centrex_id":"30",
        "display_id":"123002",
        "forwarder_list":[

        l,
        "id":"123002"
        "forwarder_list":[

        l,
        "id":"123002"
},
        "duration":5,
        "reason":"blind transfer",
        "reason_code":null,
        "state":"terminated",
        "type":"incoming"
}
}
```

Notification example for the transfer target:



```
"state":"trying",
       "type": "incoming"
"action": "update",
"result":{
       "call":{
       "callee":{
           "centrex_id":"30",
           "display_id":"123003",
"forwarder_list":[
       "caller":{
           "display_id":"123002",
       },
"connect_time":"2018-11-29 15:33:08",
"state":"connected",
       "type": "incoming"
```

Notification example for the transferee:



```
},
    "caller":{
        "account_id":"123002",
        "centrex_id":"30",
        "display_id":"123002",
        "forwarder_list":[

        ],
        "id":"123002"
      },
        "connect_time":"2018-11-29 15:33:00",
        "state":"connected",
        "type":"outgoing"
      }
}
```

join_calls

This method enables an agent to join calls when performing attended call transfer.

PortaSIP® disconnects the transferor identified by the **call** and **to_call** attributes in the call dialogs established with it and joins the remaining parties.

Parameters: JoinCallsRequest Return value: JoinCallsResponse

Realm: administrator, reseller, retail customer, account

Request example:

```
{
    "cseq":4,
    "auth_info":{
        "session_id":"e6816cb2c29d4bc9db622f5ddc0796fe"
    },
    "service":"CallControl",
    "method":"join_calls",
    "params":{
        "call":{
            "id":"1c4cef19-e035cfcd@192.168.233.134",
            "tag":"670946a915d2dd7do1"
        },
        "to_call":{
            "id":"hDSjA!Ei3bMIfb6OxfLndUwXqqF9@192.168.243.133~1o

",
        "tag":"kc3kqksjw670th8h.o"
        },
    }
}
```

Response example:

```
{
    "cseq":4,
    "result":{
```



```
"success":1
},
"success":1
}
```

Type Reference

GetSipCallsListRequest structure

Property	Type	Description
i_account	unsignedLong	The unique ID of the account record. The account represents a phone line or an office extension. To get the account ID, call the get_account_list method. The <i>i_account</i> is returned in the AccountInfo structure.
		See https://www.portaone.com/docs/Porta Billing_API.html#AccountInfo.
i_main_office	unsignedLong	The unique ID of the main office (customer record with office type 3). To get the main office ID and headquarters office type, call the get_customer_info method. Possible values for the <i>i_office_type</i> attribute in the response are the following: • 1 – none • 2 – branch office • 3 – main office
		If the office type is 1 (none), leave this attribute empty. If the office type is 2 (branch office), specify the office ID from the <i>i_main_office</i> attribute. If the office type is 3 (main office), specify the main office ID from the <i>i_customer</i> attribute. See
		https://www.portaone.com/docs/Porta Billing_API.html#CustomerInfo.



i_customer	unsignedLong	The unique ID of the customer record.
		To get the customer ID, call the get_customer_info method. The <i>i_customer</i> is returned in the CustomerInfo structure.
		See https://www.portaone.com/docs/Porta Billing_API.html#CustomerInfo.
i_ivr_an	unsignedLong	The unique ID of the access number associated with the custom IVR application (it has the User application type on the PortaBilling® web interface).
		In PortaBilling® the IVR access number is also associated with the account record. This is required to apply charges for using this number. Call the obtain_access_number API method to assign the access number to the account and receive its ID in the response.
		See https://www.portaone.com/docs/Porta Billing_API.html and https://www.portaone.com/docs/Porta Billing_API.html#ObtainAccessNumber Response.
		If you operate under the customer, reseller or the administrator realm, first retrieve the <i>i_account</i> to which you wish to assign the access number and pass it in the API request.

GetSipCallsListResponse structure

Property	Type	Description
calls list	Array of	The list of calls
	SIPCallInfo	
	structure	



SIPCallinfo structure

Property	Type	Description	
	Common		
call	SipCallIdentifi	The unique ID of separate parts	
	er structure	of the call.	
callee	SipCallerInfo	The information about who the	
	structure	caller is calling to	
caller	SipCallerInfo	The information about the caller.	
	structure		
state	string	Defines the current state of the call.	
		Possible values:	
		• trying – a call is initiated	
		and an outgoing request is sent;	
		• ringing – a phone is ringing;	
		• early – early media is played;	
		• terminated – a call is disconnected;	
		• connected – a call is answered / taken-from-hold and the remote side is connected;	
		• held – a call party is connected and is put on hold. This state is returned to the party which is placed on hold.	
		holding - call part is connected and is put on hold. This state is returned to the party who places the call on hold.	
		• queued – a caller is placed to a call queue;	
		• dequeued - a caller is	



		removed from the call
		queue.
		• started – the conference
		call has started
		• terminated – the
		conference call has been
		terminated
transport_id	string	The SIP address of the call
		defined in the format IP:port
transfer_success	int	A non-zero value means that the call transfer has been successfully performed.
type	string	Defines the type of the call dialog direction in terms of UA.
		Possible values:
		outgoing - outgoing call
		dialog. UA is initiator.
		• incoming - incoming call
		dialog. UA is recipient.
	State specific	information
connect_time	dateTime	The date and time when the call was connected or put on hold. It is sent for "connected", "held", "holding" call states.
duration	int	The call length (in seconds) from
		the moment when the call was connected. It is sent for
		"terminated" call state.
reason	string	Describes the reason for
		terminated or not established calls. Is sent for "terminated"
		call state.
queue_info	SipCallQueueSt	The information about a call
	ateInfo	queue status. Is sent for "queued"
reason_code	structure int	and "dequeued" call states. The code of the reason the calls
reason_code	IIIt	ended. Is sent for "terminated" state.
start_time	dateTime	The date and time when the call
		was initiated. Is sent for "trying",
		"ringing", "early", "started" call states.
		states.



terminate_time	dateTime	The date and time when the call was finished. It is sent for
IVD .:C	• • • • • • • • • • • • • • • • • • • •	"terminated" call states
IVR notificat	ion specific, shoul ca	ld never happen in regular API
digit	string	The detected DTMF digit.
		Possible values:
		• "0"
		• 1
		• "3"
		• "4"
		• "5"
		• "6"
		• "7"
		• "8"
		• "9"
		• "*"
1, 6 1		• "#" DTMF duration in milliseconds
dtmf_duration	int	
event	string	The event related to asynchronous IVR notification.
		 play_prompt_complete d – prompt playback finished; dtmf_digit_detected – single DTMF digit
		received.
order	int	The prompt playback order (2 or 3). Represents relative position of one prompt to another (if any) in playback stack.
		Possible values:



	•	"2"
	•	"3"

SipCallerInfo structure

SipCallerInfo structure reflects information passed in the PortaOne-Calling-Party, PortaOne-Redirecting-Party, PortaOne-Called-Party RADIUS attributes. See the **External System Interfaces Guide** for details.

Property	Type	Description
account_id	string	The phone number (PIN) of the party who is making the call. It is represented as an account in PortaSwitch® and is unique in the environment associated with call participant. See ID in https://www.portaone.com/docs/Porta Billing_API.html#AccountInfo.
centrex_id	string	The ID of the IP Centrex environment the party who is making the call belongs to. It is represented as a customer (the main office) in PortaSwitch®. See https://www.portaone.com/docs/Porta Billing_API.html#CustomerInfo
display_id	string	The display number provided by the callee/caller. It is taken from the From header and typically displayed on the called party's phone display.
display_name	string	The display name provided by the callee/caller. It is taken from the CLN field.
extension_id	string	The extension number configured on the PBX / within the IP Centrex environment and associated with the phone line (account ID).
huntgroup_id	string	The huntgroup number, on behalf of which the call happened.
id	string	The phone number of the calling / called party depending on whether it is an incoming or an outgoing call
forwarder_lis t	Array of SipForward erAccountI	The list of account IDs that initiated the forward.



	nfo structure	
access_numb	string	The IVR access number that receives
er		the call.

SipCallIdentifier structure

Property	Type	Description
tag	string	The call remote tag
<u>id</u>	string	The unique call identifier

SipCallQueueStateInfo structure

Property	Type	Description
i c queue	usingnedLong	The unique ID of a call queue record.
		To retrieve the <i>i_c_queue</i> , call the
		get_callqueue_list method.
		See
		https://www.portaone.com/docs/PortaBilli ng_API.html#CQInfo
position	int	A caller's position in the queue.
operators	int	The number of active operators.

SipForwarderAccountInfo structure

Property	Type	Description
<u>id</u>	string	The phone number of the party who
		initiated the forward.
display_id	string	The Display ID provided by the
		forwarder.
display_name	string	The display name.
centrex_id	string	The Centrex identifier.
extension_id	string	The extension ID.
huntgroup_id	string	The ID of the huntgroup involved in
	_	establishing the call.

AnswerCallRequest structure

Property	Type	Description
<u>callee_answer_mode</u>	string	Specifies the exact method
		to use for call answering.
		Possible values:
		Notify – PortaSIP®
		sends NOTIFY
		request with "Event:
		talk" (see
		BroadWorks Remot



		e Control Talk Event Package).
		Invite – PortaSIP® reconnects the call using new dialog and "auto-answer" header for INVITE:
		Alert-Info: <sip:127.0.0.1>;info= AutoAnswer Call-Info: <sip:127.0.0.1>;answe r-after=0</sip:127.0.0.1></sip:127.0.0.1>
call	SipCallIdentifier	The unique IDs of separate
	structure	parts of the call.

AnswerCallResponse structure

Property	Type	Description
success	int	A non-zero value means that the
		operation was completed.

TerminateCallRequest structure

Property	Type	Description
<u>call</u>	SipCallIdentifier	The unique IDs of separate
	structure	parts of the call

TerminateCallResponse structure

Property	Type	Description
success	int	A non-zero value means that the
		operation was completed

HoldCallRequest structure

Property	Type	Description
call	SipCallIdentifier	The unique ID of separate
	structure	parts of the call.

HoldCallResponse structure

Property	Type	Description
success	int	A non-zero value means that the
		operation was completed.



UnholdCallRequest structure

Property	Type	Description
<u>call</u>	SipCallIdentifier	The unique ID of separate
	structure	parts of the call

UnholdCallResponse structure

Property	Type	Description
success	int	A non-zero value means that the
		operation was completed.

TransferCallRequest structure

Property	Type	Description
<u>call</u>	SipCallIdentifier	The unique ID of the call
	structure	party that initiates transfer
		(acts as transferor)
<u>cld</u>	string	The phone number of the
		transfer target
sip_headers	SipHeaderInfo	The set of SIP headers.
	structure	

TransferCallResponse structure

Property	Type	Description
success	int	A non-zero value means that the
		operation was completed

SipHeaderInfo structure

Property	Type	Description
<u>name</u>	string	The header name.
value	string	The header value.

JoinCallsRequest structure

Property	Type	Description
call	SipCallIdentifier	The identifier of the call party
	structure	to be joined
to call	SipCallIdentifier	The identifier of the call party
	structure	to be joined with

JoinCallsResponse structure

Property	Type	Description
success	int	A non-zero value means that the
		operation was completed.



OriginateAdvancedCallRequest structure

Property	Type	Description
bill_id	string	The account number to be charged for a call
callee_auto_pickup	string	This flag indicates whether to request auto-answer functionality from the caller's user agent. When enabled, PortaSIP® sends the INVITE request to establish the call, which contains "auto-answer" header fields:
		Alert-Info: <sip:127.0.0.1>;info=AutoAnsw er Call-Info: <sip:127.0.0.1>;answer- after=0</sip:127.0.0.1></sip:127.0.0.1>
		The UA should support this functionality.
		Possible values:
		Y - enable auto-answer functionality;
		N - disable auto-answer
		functionality (default value).
callee id	string	The phone number to be called
caller_id	string	The phone number of the calling party

OriginateAdvancedCallResponse structure

Property	Type	Description
call	SipCallIdentifier	The unique ID of the
	structure	originated call.
success	int	A non-zero value means that
		the operation was completed.



Call control API for IVR

URL (namespace): wss://portabillingweb.yourdomain.com/ws/CallControl

These API methods enable development and operation of custom IVR applications. Thus, users can manage the incoming calls and automate their processing (e.g. to play specific prompt upon user DTMF input).

Methods

play_prompt

Use this method to play an IVR prompt to a user for a call to the IVR access number. The prompts are stored on the remote server (separately or together with the IVR application). PortaSIP® must have access to this server.

Parameters: PlayPromptRequest Return value: PlayPromptResponse

Realm: administrator, reseller, retail customer, account

Request example:

```
{
    "cseq":5,
    "auth_info":{
        "session_id":"0160ec8ffb4a394908d8be40e5b013d4"
    },
    "service":"CallControl",
    "method":"play_prompt",
    "params":{
        "call":{
            "id":"3300f0b3-7a54a006@192.168.233.134",
            "tag":"100021"
        },
        "order":3,
        "repeat":-1,
        "url":"http://192.168.233.137:8080/files/welcome",
     },
}
```

Response example:

```
{
    "cseq":5,
    "result":{
        "success":1
    },
    "success":1
}
```



stop_play_prompt

Use this method to stop playing prompts for a call to User Application access number.

Parameters: **StopPlayPromptRequest**Return value: **StopPlayPromptResponse**

Realm: administrator, reseller, retail customer, account

Request example:

```
{
    "cseq":7,
    "auth_info":{
        "session_id":"0160ec8ffb4a394908d8be40e5b013d4"
    },
    "service":"CallControl",
    "method":"stop_play_prompt",
    "params":{
        "call":{
            "id":"3300f0b3-7a54a006@192.168.233.134",
            "tag":"100021"
        },
        "order":3,
    },
}
```

Response example:

```
{
    "cseq":7,
    "result":{
        "success":1
    },
    "success":1
}
```

Notification example for the IVR access number:



```
"id":"123002"
},

"connect_time":"2018-12-05 08:51:16",

"event":"play_prompt_completed",

"order":2,

"state":"connected",

"type":"incoming"
}
}
}
```

start_dtmf_detect

Use this method to start detecting user DTMF inputs for a call to the IVR access number.

PortaSIP® supports the following DTMF modes: inband, RFC2833 and SIP INFO.

Parameters: StartDtmfDetectRequest Return value: StartDtmfDetectResponse

Realm: administrator, reseller, retail customer, account

Request example:

```
{
    "cseq":4,
    "auth_info":{
        "session_id":"0160ec8ffb4a394908d8be40e5b013d4"
    },
    "service":"CallControl",
    "method":"start_dtmf_detect",
    "params":{
        "call":{
            "id":"3300f0b3-7a54a006@192.168.233.134",
            "tag":"100021"
        },
    }
}
```

Response example:

```
{
    "cseq":4,
    "result":{
        "success":1
    },
    "success":1
}
```

Notification example for the DTMF received from the user:

```
{
    "action":"update",
    "event":"sip.call_control_notifications",
    "result":{
```



```
"call_info":{
    "call":{
        "id":"3300f0b3-7a54a006@192.168.233.134",
        "tag":"100021"
    },
    "callee":{
        "access_number":"4567890"
    },
    "caller":{
        "account_id":"123002",
        "centrex_id":"30",
        "display_id":"123002",
        "forwarder_list":[

        ],
        "id":"123002"
    },
    "connect_time":"2018-12-05 08:51:16",
        "digit":"1",
        "ddmf_duration":220,
        "event":"dtmf_digit_detected",
        "state":"connected",
        "type":"incoming"
    }
}
```

stop_dtmf_detect

Use this method to stop detecting DTMF inputs for a call to User Application access number.

Parameters: StopDtmfDetectRequest Return value: StopDtmfDetectResponse

Realm: administrator, reseller, retail customer, account

Request example:

```
{
    "cseq":17,
    "auth_info":{
        "session_id":"0160ec8ffb4a394908d8be40e5b013d4"
    },
    "service":"CallControl",
    "method":"stop_dtmf_detect",
    "params":{
        "call":{
            "id":"e0c8e7c-80b037c9@192.168.233.134",
            "tag":"100022"
            },
        },
     },
}
```

Response example:

```
{
"cseq":17,
"result":{
```



```
"success":1
},
"success":1
}
```

ring_call

Use this method to inform the caller about the call's progress. PortaSIP® sends the "180 Ringing" response.

Parameters: RingCallRequest Return value: RingCallResponse

Realm: administrator, reseller, retail customer, account

Request example:

```
{
   "cseq": 2,
   "auth_info": {
        "session_id": "649d127e95calae62ec735df99ebf01c"
   },
   "service": "CallControl",
   "method": "ring_call",
   "params": {
        "call": {
            "id": "1202668471@192.168.233.48",
            "tag": "100007"
        }
   }
}
```

Response example:

```
{
    "cseq":2,
    "result":{
        "success":1
    },
    "success":1
}
```

Notification example for the IVR application:



progress_call

Use this method to inform the caller about the call's progress. PortaSIP® sends the "183 Early media" response. The caller can hear a custom ring tone.

Parameters: ProgressCallRequest Return value: ProgressCallResponse

Realm: administrator, reseller, retail customer, account

Request example:

```
{
    "cseq": 2,
    "auth_info": {
        "session_id": "649d127e95ca1ae62ec735df99ebf01c"
    },
    "service": "CallControl",
    "method": "progress_call",
    "params": {
        "call": {
            "id": "499624331@192.168.233.48",
            "tag": "100010"
        }
    }
}
```

Response example:

```
{
    "cseq":2,
    "result":{
        "success":1
    },
    "success":1
}
```

Notification example for the IVR application:

```
{
    "action":"update",
    "event":"sip.call_control_notifications",
    "result":{
        "call_info":{
```



```
"call":{"id":"499624331@192.168.233.48",
    "tag":"100010"
},

"callee":{
        "access_number":"*51"
},
    "caller":{
            "account_id":"123003",
            "centrex_id":"30",
            "display_id":"123003",
            "display_name":"123003",
            "forwarder_list":[],
            "id":"123003"
},
        "start_time":"2019-06-07 11:07:08",
        "state":"early",
        "type":"incoming"
}
}
```

Type Reference

PlayPromptRequest structure

Property	Type	Description
<u>call</u>	SipCallIdentifier	The unique ID of separate parts
	structure	of the call
url	string	The URL path to prompt files. Supported protocols are HTTP, HTTPS. Supported audio formats are: au, g729. The .au file must be in 8-bit G.711 u-law data encoding format.
		The URL must contain path to the file without file extension. E.g. if prompts are 'http://myhost.com/prompt.au', 'http://myhost.com/prompt.g72 9', url is 'https://myhost.com/prompt'.
		It is not mandatory to have prompts in all formats.
order	int	Prompt playback order (2 or 3). Represents relative position of one prompt to another (if any) in playback stack.



		Default value: 2
repeat	int	The number of times to repeat the prompt. "-1" - repeat playback forever. Default value: 1

PlayPromptResponse structure

Property	Type	Description
success	int	A non-zero value means that the
		operation was completed

StartDtmfDetectRequest structure

Property	Type	Description
call	SipCallIdentifier	The unique ID of separate
	structure	parts of the call

StartDtmfDetectResponse structure

Property	Type	Description
success	int	A non-zero value means that the
		operation was completed

StopDtmfDetectRequest structure

Property	Type	Description
<u>call</u>	SipCallIdentifier	The unique IDs of separate
	structure	parts of the call

StopDtmfDetectResponse structure

Property	Type	Description
success	int	A non-zero value means that the
		operation was completed

StopPlayPromptRequest structure

Property	Type	Description
<u>call</u>	SipCallIdentifier	The unique ID of separate
	structure	parts of the call
order	int	Prompt playback order (2 or
		3). Represents relative
		position of one prompt to
		another (if any) in playback
		stack.



StopPlayPromptResponse structure

Property	Type	Description
success	int	A non-zero value means that the
		operation was completed

RingCallRequest structure

Property	Type	Description
<u>call</u>	SipCallIdentifier	The identifier of the call
	structure	

RingCallResponse structure

Property	Type	Description
success	int	A non-zero value means that the
		operation has been successfully
		completed

ProgressCallRequest structure

Property	Type	Description
<u>call</u>	SipCallIdentifier	The identifier of the call
	structure	

ProgressCallResponse structure

Property	Type	Description
success	int	A non-zero value means that the
		operation has been successfully
		completed

Conferencing API

These API methods enable agents to effectively manage their calls by converting active calls into conferences on the fly. While in a conference, agents can add / remove, mute and / or hold participants (e.g. to have a private talk with some participants while others remain connected). An agent can leave and join the conference they created at any moment while the call for the remaining participants continues.

An agent is represented as an account in PortaSwitch®. When the API session is established under the account realm, the agent who creates a conference automatically becomes its owner and can manage only this conference. When the API session is established under the customer realm, the agent has access to all conferences in the IP Centrex



environment and can manage them (e.g. to appoint another account as the conference owner.) A conference owner is charged for the conferencing service.



To create conference rooms and / or convert calls into conferences, the agent's product configuration must include both the Voice calls and Conferencing services.

Methods

join_on_spot_conference

Use this method to create a conference room or to join to an active conference. When called under the account realm, this method enables the agent to join only the conference they have created. When called under the customer realm, this method enables creating conferences on behalf of any account under the customer and joining any active conference in the IP Centrex environment.

Parameters: JoinOnSpotConferenceRequest
Return value: JoinOnSpotConferenceResponse
Realm: administrator, reseller, retail customer, account

Request example:

Response example:



```
{
  "cseq":2,
    "success": 1,
  "call": {
    "id": "cSsJ!DNMIEZfxJrZlKJja12vKcb4@192.168.233.134"
  }
}
```

Notification example about the conference sent to the conference owner:

```
{
   "action":"update",
   "event": "sip.call_control_notifications",
   "result": {
        "conference_info": {
            "id": "cSsJ!DNMIEZfxJrZlKJja12vKcb4@192.168.233.134",
            "tag": 100028
        },
        "conference_name": "test's conference",
        "conference_owner_id": "12065550021",
        "event": "ConferenceInfo",
        "start_time": "2019-02-01 13:15:55",
        "status": "started"
        }
    }
}
```

Notification example about the participant sent to the conference owner:



convert_to_on_spot_conference

Use this method to convert an active call to a conference call. The agent's product configuration in PortaBilling® must include the Conferencing service.

Parameters: ConvertToOnSpotConferenceRequest Return value: ConvertToOnSpotConferenceResponse Realm: administrator, reseller, retail customer, account

Request example:

```
"cseq":2,
    "method": "convert to on spot_conference",
    "params": {
      "callee": {
        "moderator": "N",
        "max participants": "15",
        "lang": "en",
"quiet_mode": "N",
        "music on hold url":
        "announced_conference_name_url": "
http://91.212.119.34:5687/test conf",
      "wait for moderator": "Y"
    },
"caller": {
       "moderator": "Y",
       "max_participants": "15",
       "lang": "en",
"http://192.168.233.134:5687/test_conf"
      "name": "_test "
      "tag": "fd15be61f2813a4eo2",
      "id": "f424a21d-908b49c2@192.168.***.***"
    "session id": "637efe9391d6a9b109610<u>52e861</u>19655"
```

Response example:

```
{
    "cseq":2,
    "result":{
        "success":1
    },
    "success":1
}
```



get_on_spot_conference_list

Use this method to query the list of currently active conferences and receive the information about them. For this, subscribe to API notifications using the **enable_api_notifications** method.

Parameters: GetOnSpotConferenceListRequest Return value: GetOnSpotConferenceListResponse Realm: administrator, reseller, retail customer, account

Request example:

```
"cseq": 3,
   "auth_info": {
        "session_id": "637efe9391d6a9b10961052e86119655"
    },
        "service": "CallControl",
        "method": "get_on_spot_conference_list",
        "params": {
            "conference_info": {}
}
```

Response example:

get_on_spot_conference_participant_list

Use this method to query information about participants in a given conference. The response contains all participants: those who are in the conference and those who left.

Parameters: GetOnSpotConferenceParticipantListRequest Return value: GetOnSpotConferenceParticipantListResponse Realm: administrator, reseller, retail customer, account

Request example:

```
"cseq": 3,
    "auth_info": {
        "session_id": "637efe9391d6a9b10961052e86119655"
     },
     "service": "CallControl",
     "method": "get_on_spot_conference_participant_list",
     "params": {
```



```
"conference_info": {
        "name":"test_conference",
        "owner_id": "12065550021"
    }
}
```

Response example:

update_on_spot_conference_participant

Use this method to update the participant status. This method enables you to mute and unmute participants, put them on hold and release them from hold.

Parameters: UpdateOnSpotConferenceParticipantRequest Return value: UpdateOnSpotConferenceParticipantResponse Realm: administrator, reseller, retail customer, account

Request example (to mute a participant):

```
{
    "cseq":2,
        "service": "CallControl",
    "method": "update_on_spot_conference_participant",
    "params": {
        "participant_info": {
             "participant_id": "1",
             "mute": "Y",
             "music_on_hold_url":
"http://91.212.119.34:5687/8_bit",
             "hold": "N",
        },
        "conference_info": {
             "name": "test"
             "owner_id":"12065550021"
        },
        "auth_info": {
             "session_id": "637efe9391d6a9b10961052e86119655"
```



```
}
}
```

Response example:

```
{
    "cseq":2,
        "result":{
        "success":1
    },
    "success":1
}
```

delete_on_spot_conference_participant

Use this method to delete participants from on the spot conference by disconnecting them.

Parameters: DeleteOnSpotConferenceParticipantRequest Return value: DeleteOnSpotConferenceParticipantResponse Realm: administrator, reseller, retail customer, account

Request example:

Response example:

```
{
    "cseq":2,
        "result":{
        "success":1
    },
    "success":1
}
```



Type reference

JoinOnSpotConferenceRequest structure

Property	Type	Description
call info	SipOnSpotConference	The information about
	CallInfo structure	the call
conference_info	SipOnSpotConference	The information about
	Info structure	the conference
participant info	SipOnSpotConference	The information about
	ParticipantInfo ParticipantInfo	the conference participant
	structure	

JoinOnSpotConferenceResponse structure

Property	Type	Description
success	int	A non-zero value means that
		the operation was completed
call	SipCallIdentifier	The unique ID of the
	structure	originated call.

SipOnSpotConferenceCallInfo structure

Property	Type	Description
caller id	string	The phone number used to
		establish a call to a new
		conference participant
<u>callee id</u>	string	The phone number of a
		conference participant to be
		called
id	string	The call identifier
sip_auth_username	string	The SIP username used for
		authorization and charges for
		a call to a new conference
		participant. By default, the
		conference room owner is
		charged for a voice call to a
		conference participant and
		the time spent by the
		participant in the conference
		room.
		It is equal to the owner_id
		attribute if the method is
		executed under the account
		realm.
		If the method is executed



		under the customer realm,
		this attribute's value must
		correspond to IDs of one of
		the accounts of this customer
tag	string	The call remote tag
transport_id	string	The SIP transport ID

SipOnSpotConferenceInfo structure

Property	Type	Description
name	string (max 64 chars)	The name of the conference room. It must be unique among the rooms created by the agent or within the IP Centrex environment. This means that any user who joins "room3121" under specific account/customer credentials will end up in the same conference room, while the user connecting through different accounts/customers will not
owner id	string	The unique ID of the conference room owner's account. By default, and if the <i>sip_auth_username</i> attribute is not defined, the conference room owner is charged for voice calls to conference participants and the time spent by every participant in the conference room. If the method is executed under account realm, this value is not mandatory as it is equal to this account ID. If executed
i_customer	int	under the customer realm, this attribute's value must correspond to IDs of one of the accounts of this customer The unique ID of the customer record. To get the customer ID, call the get_customer_info method. The i_customer is



returned in the CustomerInfo
structure.
See
https://www.portaone.com/docs/Porta
Billing_API.html#CustomerInfo.

SipOnSpotConferenceParticipantInfo structure

Property	Type	Description
Property announced_conference_name_url	Type string	Specifies a URL for the sound file with the name of the conference which will be announced to people joining it. Supported protocols are HTTP, HTTPS. Supported audio formats are: au, g729. The .au file must be in the 8-bit G.711 u-law data encoding format. The URL must contain the path to the file without the file extension. E.g. if the prompts are
		'http://myhost.com/prompt.au', 'http://myhost.com/prompt.g729', the URL is 'https://myhost.com/prompt'.
		It is not mandatory to have prompts in all formats.
caller_id	string	The phone number used to establish a call to a new conference participant
hold	string	This flag indicates whether the participant is on hold in the conference. Possible values: Y, N
join_time	date Time	The date and time when the participant joined the on-the-spot conference
teave_time	date	The date and time when



	Time	the participant left the on- the-spot conference
play_announce	string	The flag shows whether to play an announcement before removing a participant from the conference
lang	string	The ISO 639-1 language the user prefers when interacting with the conferencing IVR application. Find the list of supported languages in Appendices section of the PortaSIP® Media Applications Guide. If the specified language is unsupported, PortaSIP® plays prompts in English. Default value: en
max_participants	int	The maximum number of concurrent participants allowed in the conference room. This value must not exceed the maximum number of simultaneous participants defined for the conference room owner within their account or product configuration
moderator	string	This flag indicates whether the participant is the moderator. A moderator is the person who is responsible for arranging the conference and is usually the host or leader of the conference. Possible values: Y, N Default value: N
music_on_hold_url	string	Specifies a URL for the sound file with the name of the conference which will be announced to people joining it.



		Supported protocols are HTTP, HTTPS. Supported audio formats are: au, g729. The .au file must be in the 8-bit G.711 u-law data encoding format. The URL must contain the path to the file without the file extension. E.g. if the prompts are
		'http://myhost.com/prom pt.au', 'http://myhost.com/prom pt.g729', the URL is 'https://myhost.com/pro
		mpt'.
		It is not mandatory to have prompts in all formats.
mute	string	The flag indicates whether the participant can speak in the conference.
		Possible values: Y, N Default value: N
quiet_mode	string	This flag indicates whether PortaSIP® plays the welcome message and enter / leave sounds to conference participants. When enabled, the participant joins the conference without others being aware of that (e.g. to monitor the conversation.) Possible values: Y, N
		Default value: N
participant_id	int	The unique identifier of the conference participant
wait_for_moderator	string	This flag indicates that the conference participants will not be able to communicate with each



other until a moderator joins the room. When this flag is enabled and the last moderator leaves the conference room, the conference ends and all other participants are
disconnected.
Possible values: Y, N.
Default value: N

ConvertToOnSpotConferenceRequest structure

Property	Type	Description
conference_info	SipOnSpotConferenceInfo	The
	structure	information
		about the
		conference
call info	SipOnSpotConferenceCallInfo	The
	structure	information
		about the
		call
<u>callee</u>	SipOnSpotConferenceParticipantI	The
	nfo structure	information
		about the
		participant
		who the
		caller was
		calling to
<u>caller</u>	SipOnSpotConferenceParticipantI	The
	nfo structure	information
		about the
		participant
		who was
		calling

ConvertToOnSpotConferenceResponse structure

Property	Type	Description
success	int	A non-zero value means that
		the operation was completed

GetOnSpotConferenceListRequest structure

Property	Type	Description
conference info	SipOnSpotConferenceInfo	The information
	structure	about the conference



GetOnSpotConferenceListResponse structure

Property	Type	Description
conference_list	Array of	The list of active
	SipOnSpotConferenceInfo	conference rooms
	structures	

GetOnSpotConferenceParticipantListRequest structure

Property	Type	Description
conference info	SipOnSpotConferenceInfo	The information
	structure	about the conference

GetOnSpotConferenceParticipantListResponse structure

Property	Type	Description
participant_list	Array of	The information about
	SipOnSpotConferencePa	the conference
	rticipantInfo structures	participants

UpdateOnSpotConferenceParticipantRequest structure

Property	Type	Description
conference_info	SipOnSpotConferenc	The information about
	eInfo structure	the conference
participant_info	SipOnSpotConferenc	The information about
	eParticipantInfo	the conference participant
	structure	

UpdateOnSpotConferenceParticipantResponse structure

Property	Type	Description
success	int	A non-zero value means that the
		operation was completed

DeleteOnSpotConferenceParticipantRequest structure

Property	Type	Description
conference info	SipOnSpotConferenc	The information about
	eInfo structure	the conference
participant info	SipOnSpotConferenc	The information about
	eParticipantInfo	the conference participant
	structure	

DeleteOnSpotConferenceParticipantResponse structure

Property	Type	Description
success	int	A non-zero value means that the
		operation was completed



4. Appendices



Appendix A. Sample script for PortaSIP® media server SOAP communication

```
#!perl -w
use strict;
# to enable client-side script debugging uncomment the line below
# and comment the one next to it
#use SOAP::Lite +trace => 'debug';
use SOAP::Lite;
use MIME::Entity;
use Data::Dumper;
# If the server certificate is not trusted (e.g. it was not issued by a
# trusted certificate authority), then ignore it.
$ENV{PERL LWP SSL VERIFY HOSTNAME}=0;
my $soap sess = SOAP::Lite
    -> uri('https://localhost/UM/SOAP/Session')
    -> proxy('https://pum-host:8443/soap.fcgi')
    -> on fault( sub {
       \overline{my} ($soap, $res) = @_; print ("SOAP error:". (ref $res ? $res->faultstring : $soap-
>transport->status . "/" . $res));
       });
my $soap test = SOAP::Lite
    -> uri('https://localhost/UM/SOAP/Voicemail')
    -> proxy('https://pum-host:8443/soap.fcgi')
    -> on fault( sub {
       my(\$soap, \$res) = @;
       print ("SOAP error:". (ref $res ? $res->faultstring : $soap-
>transport->status . "/" . $res));
       });
my $soap dial dir = SOAP::Lite
    -> uri('https://localhost/UM/SOAP/DialDirectory')
    -> proxy('https://pum-host:8443/soap.fcgi')
    -> on fault( sub {
       \overline{my} ($soap, $res) = 0_;
       print ("SOAP error:". (ref $res ? $res->faultstring : $soap-
>transport->status . "/" . $res));
       });
my $soap aa = SOAP::Lite
    -> uri('https://localhost/UM/SOAP/AutoAttendant')
    -> proxy('https://pum-host:8443/soap.fcgi')
    -> on fault( sub {
       \overline{\text{my}}($soap, $res) = 0_;
       print ("SOAP error: ". (ref $res ? $res->faultstring : $soap-
>transport->status . "/" . $res));
        });
my $authInfo = $soap sess->login({
        'login' => '88881',
                   => 'pum.somedomain.com',
        'domain'
        'password' => 'test123'})->result();
$authInfo = SOAP::Header->name( 'auth info' => {
                'session id' => $authInfo->{'session id'}
               });
my $authInfo nosess = SOAP::Header->name( 'auth info' => {
        'login' => '88881',
        'domain'
                   => 'pum.somedomain.com',
        'password' => 'test123'
               });
```



```
#example of accessing SOAP module without establishing session
$res = $soap test->get vm settings($authInfo nosess)->result();
print Dumper($res);
$res = $soap_test->set_vm_settings($authInfo,
        {'vm_settings' =>
                'password' => '777',
               'ext email' => 'sergey.pavlov@gmail.com',
               'auto play' => 'no',
                'announce dt'=> 'no'
           } })->result();
print "set_vm_settings done\n";
my $ent = MIME::Entity->build(
                              => 'wellcome.au',
                'Filename'
               'Type' => 'audio/basic',
               'Encoding' => 'base64',
               'Path'
                          =>
'/var/lib/psmsc/prompts/en/personal ivr/frw select order.au',
my @parts = (\$ent);
$res = $soap test->parts([ $ent ])->set vm greeting($authInfo,
        {'greeting_info' =>
               'greeting type' => 'name',
               'filename'=> 'wellcome.au'
            } ))->result();
print "set vm greeting done\n";
$res = $soap test->get vm greeting($authInfo,
               'greeting type' => 'name',
           }) ->result();
print "get vm greeting done\n";
$res = $soap dial dir->get dir info($authInfo)->result();
print "get_directory_list done\n";
print Dumper($res);
$res = $soap dial dir->parts(@parts)->create dir entry($authInfo,
    'dir_entry_info' => {
                     => 'Y'.
    'active'
    'abbreviated number'=> '1787896',
   'number_to_dial' => '111111',
'lastname' => 'LName',
'description' => 'desc foo',
                      => 'wellcome.au'
    'prompt'
    } )->result();
print "create dir entry done\n";
print Dumper($res);
my $i entry = $res->{'i entry'};
$res = $soap dial dir->parts(@parts)->update dir entry($authInfo,
    'dir entry info' => {
    'i_entry' => $i_entry,
    'active'
                    => 'Y'
    'abbreviated number'=> '99',
    'number_to_dial' => '565656',
'lastname' => 'LastName',
    'lastname'
    'description' => 'desc333',
'prompt' => 'wellcome.au'
    } )->result();
```



```
print "update dir entry done\n";
$res = $soap dial dir->get dir entry($authInfo, {'i entry' => $i entry}
)->result();
print "get dir entry done\n";
print Dumper($res);
$res = $soap dial dir->del dir entry($authInfo,
 {'i_entry' => $res->{'dir_entry_info'}->{'i_entry'}})->result();
print "del dir entry done\n";
print Dumper($res);
####### Auto Attendant ##############################
$res = $soap aa->get menu list($authInfo)->result();
if (!$res) {
   print "get menu list failed\n";
print "get_menu_list done\n";
print Dumper($res);
my $root_i_menu;
foreach my $menu (@{$res->{'menu list'}}) {
   if ($menu->{'name'} eq 'ROOT') {
        $root i menu = $menu->{'i menu'};
        last;
}
$res = $soap aa->parts(@parts)->set menu prompt($authInfo,
       'i_menu'
    }) ->result();
print "set menu prompt done\n";
print Dumper($res);
$res = $soap aa->set menu transition($authInfo,
        'transition info' => {
        'i_menu' => $root_i_menu,
        'event' => '0',
'action' => 'Transfer',
        'event'
        'destination'=> '5555',
    }) ->result();
print "set_menu_transition done\n";
print Dumper($res);
$res = $soap aa->get menu transition list($authInfo,
       'i menu' => $root i menu,
    })->result();
print "get menu transition list done\n";
print Dumper($res);
$res = $soap aa->get menu prompt($authInfo,
       'i_menu' => $root_i_menu,
'prompt_type' => 'intro',
    })->result();
print "get menu prompt done\n";
print Dumper($res);
$res = $soap_aa->create_menu($authInfo,
    { 'menu_info' => {
       'name' => 'AABBBCCC',
'period' => 'hr{0-11}',
'period desc' => 'Some period',
        'msg_timeout_type' => 'standard'
    }) ->result();
print "create menu done\n";
```



```
print Dumper($res);
my $new i menu=$res->{'i menu'};
$res = $soap aa->update menu($authInfo,
    { 'menu info' => {
                          => $new i menu,
        'i menu'
        'i_menu' => $new_i_men
'name' => 'DDDEEFF',
'period' => 'hr{0-2}',
                         => 'DDDEEFF',
        'period_desc' => 'New period',
    }) ->result();
print "update menu done\n";
print Dumper(\overline{\$}res);
$res = $soap aa->get menu list($authInfo)->result();
print "get menu list done n";
print Dumper ($res);
$res = $soap aa->del menu($authInfo,
    { 'i menu' => $new_i_menu })->result();
print "del menu done\n";
print Dumper($res);
$res = $soap_sess->logout($authInfo)->result();
print "logout done\n";
print Dumper($res);
```

Appendix B. Call control API usage examples

Below you will finds scripts written in Python and JavaScript that show the communication with PortaSwitch® via the Call control API. Both scripts initiate a WebSocket connection and enable the receipt of call state notifications for the IP Centrex environment.

JavaScript

This is the .html page with the options to connect to PortaSwitch via WebSocket and start receiving call state notifications and close the connection. The action log is shown in the output field.

```
<!DOCTYPE html>
<h+m1>
<head>
    <meta charset="utf-8">
    <title>PortaBilling Call Controll API example</title>
    <script language="javascript" type="text/javascript">
      // web socket server URL
      // replace <web-server.yourdomain.com> with the actual hostname
of your web server
      // replace <port> with the required port
      var wsUri = "wss:///<web-server.yourdomain.com>:<port>/ws";
     // request body for enabling notifications receiving for the IP
Centrex
     // replace <session id> with authenticated session id (can be
obtained from Session.login response)
     // replace <i customer> with id of a customer which notifications
you want to receive
```



```
var enable notif rqst = '{"cseq": 2, "service": "Customer",
"method": "enable_api_notifications", "auth_info":
{"session id":"<session id>"}, "params": { "i customer":
"<i customer>", "event": "sip.call control notifications" }}';
      // request body for disabling notifications receiving
var disnable_notif_rqst = '{"cseq": 3, "service": "Customer",
"method": "disable_api_notifications", "auth_info":
{"session id":"<session id>"}, "params": { "i customer":
"<i customer>", "event": "sip.call control notifications" }}';
      var output;
      var websocket;
      function connect() {
          output = document.getElementById("output");
          startReceivingNotifications();
      function startReceivingNotifications() {
          // create new WebSocket instance
          websocket = new WebSocket(wsUri);
          websocket.onopen = function(evt) {
              // socket opening event handler
              onOpen(evt);
          websocket.onclose = function(evt) {
              // socket closing event handler
              onClose(evt);
          websocket.onmessage = function(evt) {
              // message receiving event handler
              onMessage(evt);
          websocket.onerror = function(evt) {
              // error event handler
              onError(evt);
          };
      }
      function onOpen(evt) {
          writeToScreen("CONNECTED");
          //send request to start notifications receiving
          doSend(enable notif rqst);
      }
      function onClose(evt) {
          writeToScreen("DISCONNECTED");
      function onMessage(evt) {
          // display received response
          writeToScreen('<span style="color: blue;">RECEIVED: ' +
evt.data + '<\/span>');
      function onError(evt) {
          // display received error message
          writeToScreen('<span style="color: red;">ERROR:<\/span> ' +
evt.data);
      }
      function doSend(message) {
          // display sent request
          writeToScreen("SENT: " + message);
          // send request to the web socket server
          websocket.send(message);
      }
      function writeToScreen(message) {
          var pre = document.createElement("p");
          pre.style.wordWrap = "break-word";
```



```
pre.innerHTML = message;
          output.appendChild(pre);
      }
      function disconnect() {
          //send request to stop notifications receiving
          doSend(disnable notif rqst);
          // close web socket
          websocket.close();
      }
    </script>
</head>
<body>
    <h2>PortaBilling Call Controll API example</h2>
    <input type="button" name="connect" value="Connect"</pre>
onclick=connect()>
    <input type="button" name="disconnect" value="Disconnect"</pre>
onclick=disconnect()>
    <div id="output" style="height:200px; border: 1px solid black;</pre>
overflow: scroll; margin-top: 10px;"></div>
</body>
</html>
```

Python

This script connects to the WebSocket server, enables the receipt of notifications, waits for 10 seconds for notifications. After that the script disables the receipt of notifications and closes the WebSocket connection.

```
import websocket
try:
    import thread
except ImportError:
    import thread as thread
import time
# web socket server URL
# replace <web-server.yourdomain.com> with the actual hostname of your
web server
ws uri = "wss://<web-server.yourdomain.com>/ws"
# request body for enabling notifications receiving for the IP Centrex
# replace <session id> with authenticated session id (can be obtained
from Session.login response)
# replace <i customer> with id of a customer which notifications you
want to receive
enable_receiving = '{"cseq": 3, "service": "Customer", "method":
"enable_api_notifications", "auth_info": {"session_id": "<session</pre>
id>"}, "params": { "i customer": "<i customer>", "event":
"sip.call_control_notifications" }}'
# request body for disabling notifications receiving
disable receiving = '{"cseq": 3, "service": "Customer", "method":
"disable_api_notifications", "auth info": {"session id": "<session
id"}, "params": { "i_customer": "<i customer>", "event":
"sip.call_control_notifications" }}'
def on message(ws, message):
    print("RECEIVED: %s" % message)
def on_error(ws, error):
    print(error)
def on_close(ws):
```



```
print("DISCONNECTED")
def on_open(ws):
    def run(*args):
        print("CONNECTED")
        print("SENT: %s" % enable receiving)
        # send request to start notifications receiving
        ws.send(enable receiving)
        # waiting for notifications
        time.sleep(10)
        # send request to stop notifications receiving
        ws.send(disable receiving)
        # wait before closing websocket
        time.sleep(5)
        # close websocket
        ws.close()
    thread.start new thread(run, ())
if __name__ == " main ":
    ws = websocket.WebSocketApp(ws_uri,
                              on message = on message, # message
receiving event handler
                              on_error = on_error,
                                                       # error event
handler
                              on close = on close)
                                                       # socket closing
event handler
    ws.on open = on open
    ws.run forever()
```

Appendix C. How to define a time period

A time period is specified as a string in the following format:

```
sub-period[, sub-period...]
```

A sub-period takes the following form:

```
scale {range [range ...]} [scale {range [range ...]}]
```

The scale must be one of the nine different options (or their equivalent codes):

Scale	Scale Code	Valid Range Values
year	yr	n – where n is an integer
		$0 \le n \le 99 \text{ or } n \ge 1970$
month	mo	1-12 or jan, feb, mar, apr, may,
		jun, jul, aug, sep, oct, nov, dec
week	wk	1-6
yday	yd	1-365
mday	md	1-31
wday	wd	1-7 or su, mo, tu, we, th, fr, sa
hour	hr	0-23 or 12am 1am-11am 12noon
		12pm 1pm-11pm



minute	min	0-59
second	sec	0-59

The same scale type may be specified multiple times. Additional scales simply extend the range defined by previous scales of the same type. The range of a given scale must be a valid value in the form:

V

or

v-v

In the range specification *v-v*, if the second value is larger than the first, the range wraps around unless the scale specification is "year". Year does not wrap because a year is never really reset, rather it just changes by increments.

Ignoring that fact that led to the dreaded Y2K nightmare, when a year rolls over from 99 to 00, it has really rolled over one century, not gone back a century. Time period supports the ambiguous two digit year notation because it is so widespread.

However, two-digit notation is converted to four digits by prepending the first two digits of the current year. In the case of 99-1972, the 99 is translated to whatever the current century is (probably the 20th), and so the range 99-1972 is treated as 1972-1999. For the 21st century, the range would then be 1972-2099.

In any case, if v-v is 9-2, and the scale is month, September, October, November, December, January, and February are the months specified by the range (9-2 is the same as Sep-Feb).

If v-v is 2-9, then the valid months are February, March, April, May, Jun, July, August, and September.

v is not a point in time. For the hour scale, 9 specifies the time period from 9:00:00 am to 9:59:59 am. This is what most people would call 9-10.

In other words, v is discrete in its time scale. 9 changes to 10 when 9:59:59 changes to 10:00:00, but 9 is the period from 9:00:00 to 9:59:59. Just before 9:00:00, v was 8.

Note that there can be a white space anywhere, and case is unimportant. Note also that scales must be specified either in long form (year, month, week, etc.) or in code form (yr, mo, wk, etc.). Scale forms in a period statement may be mixed.



Furthermore, when using letters to specify ranges, only the first two (for weekdays) or the first three (for months) are significant. January is a valid specification for Jan, and Sunday is a valid specification for su. Sun is also valid for su.

Period Examples

Example 1

To specify a time period from Monday through Friday, 9 a.m. to 5 p.m., use the following period:

```
wd {Mon-Fri} hr {9am-4pm}
```

When specifying a range using "-", it is best to think of "-" as meaning "through", i.e. 9 a.m. through 4 p.m., which is the time interval ending just before 5 p.m.

Example 2

To specify a time period from 9 a.m. to 5 p.m. on Monday, Wednesday, and Friday and from 9 a.m. to 3 p.m. on Tuesday and Thursday, use the following period:

```
wd {Mon Wed Fri} hr {9am-4pm}, wd{Tue Thu} hr {9am-2pm}
```

Example 3

To specify a time period that extends from Monday to Friday, 9 a.m. to 5 p.m., but alternates the weeks in a month, use the following period:

```
wk {1 3 5} wd {Mon Wed Fri} hr {9am-4pm}
```

Example 4

For a period that specifies the winter:

```
mo {Nov-Feb}
```

The next example is equivalent to the previous one:

```
mo {Jan-Feb Nov-Dec}
as is:
mo {jan feb nov dec}
or also:
mo {Jan Feb}, mo {Nov Dec}
and this, too:
```



mo {Jan Feb} mo {Nov Dec}

Example 5

To specify a period of every other half-hour, use something like this: minute {0-29}

Example 6

To specify the morning, use the following period definition:

```
hour {12am-11am}
```

Please note that '11 a.m.' here is not the 11:00:00 a.m. time point but the 11:00:00 a.m.-11:59:59 a.m. interval.

Example 7

To specify the period that consists of several 5-second blocks:

```
sec {0-4 10-14 20-24 30-34 40-44 50-54}
```

Example 8

To specify every first half-hour on alternating weekdays, and the second half-hour during the rest of the week, use the following period:

```
wd \{1\ 3\ 5\ 7\} min \{0-29\}, wd \{2\ 4\ 6\} min \{30-59\}
```