



User Guide

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www.telinta.com

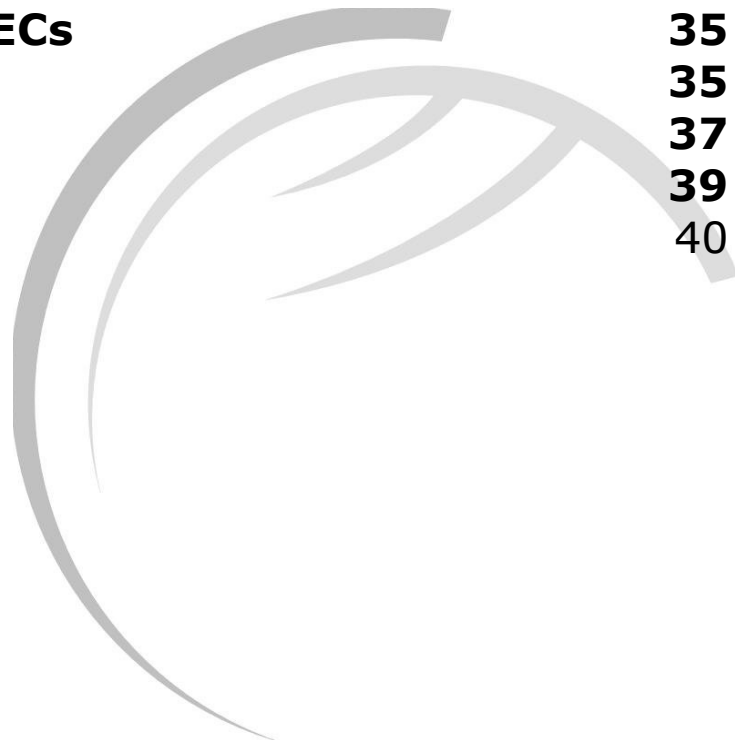
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Introduction

About Telinta

Founded in 2002, Telinta offers secure and reliable cloud-based solutions for VoIP service providers. Our full portfolio of white label solutions is highly customizable for you and your resellers. With Telinta, you can focus on growing your business, while we focus on managing your technology.

TeliCore™ is our robust hosted softswitch platform, integrating world-class carrier-grade Switching and Billing capabilities from PortaSwitch™, with Telinta's own proprietary enhancements. TeliCore is the largest hosted softswitch installation of its type, anywhere in the world.

Telinta offers:

- Cloud-based softswitch - rapid deployment, with no capital investment
- Full portfolio of ready to use turn-key solutions for VoIP service providers
- Best-of-breed VoIP technology
- Highly stable, scalable, and customizable platform
- Ability to select your own VoIP termination and origination partners
- Comprehensive training and Around-The-Clock Technical Support.

Key Features And Benefits

TeliCon audio conferencing solution includes most of commonly used in-conference touchtone and web interface audio conferencing features available in the market today.

TeliCon key benefits include:

- Full-featured Audio Conferencing integrated into Telinta's hosted platform
- Rigorous security measures to help prevent unauthorized use
- Reservationless and Scheduled audio conferences
- Moderated and non-moderated conferences
- Ability to Mute/Unmute participants
- Virtually unlimited number of conference rooms
- Customizable Music on Hold (MOH)
- Auto-generation of Moderator and participant access codes
- Announcing the number of conference room participants
- Limit on duration of the conference session
- Limit on the number of conference participants
- Ability to lock the conference room after the last participant leaves
- Brandable web interface to setup audio conferences
- Call recording with immediate download
- "Drop last participant" from the Moderator touchtone keypad
- "Drop" a selected participant via the Moderator web-interface.

Configuration Checklist

Begin by printing the checklist shown below. Follow the system setup instructions detailed in this chapter, checking off each completed operation on the printed list. Please be sure to perform all of the operations (all of the boxes must be checked), otherwise the service will not work.

General configuration	
Add a new node	<input type="checkbox"/>
Add a new destination CONF	<input type="checkbox"/>
Vendor provisioning	
Create a new tariff, add a rate for the destination CONF	<input type="checkbox"/>
Create a new vendor	<input type="checkbox"/>
Add a new vendor connection using the vendor tariff	<input type="checkbox"/>
Account provisioning	
Create a customer who owns the conference	<input type="checkbox"/>
Create a tariff with "Voice Calls" and "Conferencing" service type, add rates for the destination CONF to both tariffs	<input type="checkbox"/>
Add a new product	<input type="checkbox"/>
Add two usage charges entries for "Voice Calls" and "Conferencing" services to the product, assign them the corresponding tariffs	<input type="checkbox"/>
Create a new account that will be used as an access number, assign it the product created before	<input type="checkbox"/>
Add forwarding to SIP URI CONF@199.180.223.108	<input type="checkbox"/>
Log in to Account Self-Care interface and add a conference room	<input type="checkbox"/>
Testing	
Register several SIP phones	<input type="checkbox"/>
Call Access Number and test the conference room	<input type="checkbox"/>

Instructions and Feature Explanations

This section provides instructions to configure the conferencing service and full descriptions of the conference features. Using Telinta's convenient web interface, VoIP service providers can customize their TeliCon conferencing service. This allows you to easily control pricing, use any vendor you need for origination or termination, and utilize other key features.

General Configuration

Log into **TeliCore** as a root or admin user.



Figure 1. Logging in

TIP

Click on the **Password Recovery** button to restore the forgotten password.

1. Node configuration

Go to the **Nodes** menu.

Nodes are gateways which provide accounting information or exchange authentication and authorization information with TeliCore. On the **Node Management** page you can see a list of nodes already registered with the system.

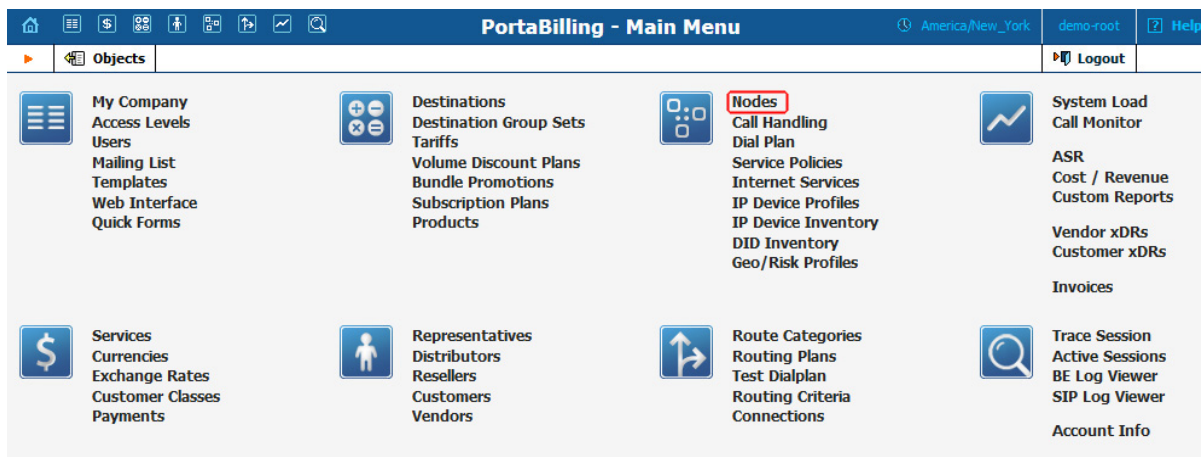


Figure 2. Nodes menu

Add a new node.

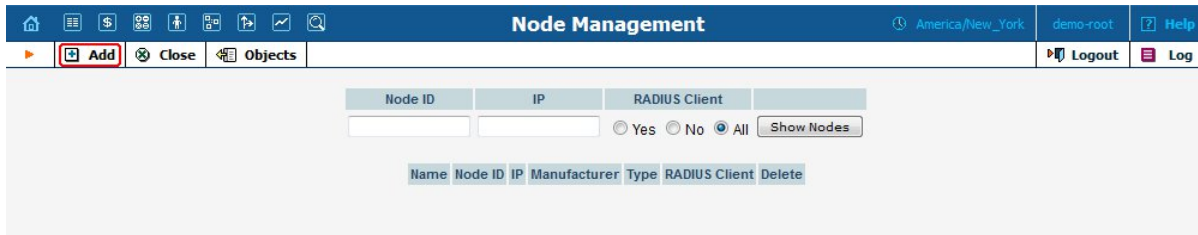


Figure 3. Adding a conferencing node

Configure your new node as follows, replacing **34** with the number of your virtual partition (see *Figure 4. Node configuration* below):

- Node Name: TeliCon + **number of your virtual partition** (e.g. TeliCon34)
- Manufacturer: PortaOne
- Type: Generic
- Node ID: TeliCon + **number of your virtual partition** (e.g. TeliCon34)
- NAS-IP-Address: 10.0.5.**34** (use number of your virtual partition instead of **34**)
- Auth. Transl. Rule: leave empty
- RTP Proxying: On NAT
- RADIUS client: Yes
- RADIUS Key: Auto
- RADIUS Source IP: 10.0.5.**34** (use number of your virtual partition instead of **34**)
- RADIUS Dictionary: Cisco.

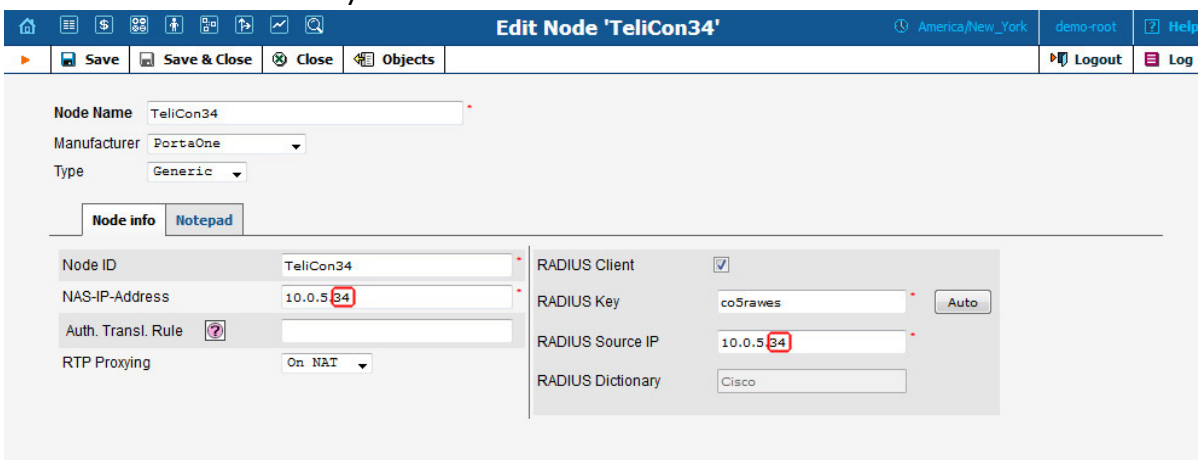


Figure 4. Node configuration

2. Adding destination CONF

Go to **Destinations** and add a new destination called CONF.



Figure 5. Entering the Destinations sub-menu

Click Add, fill in the prefix CONF, and then click on the **Save** icon.

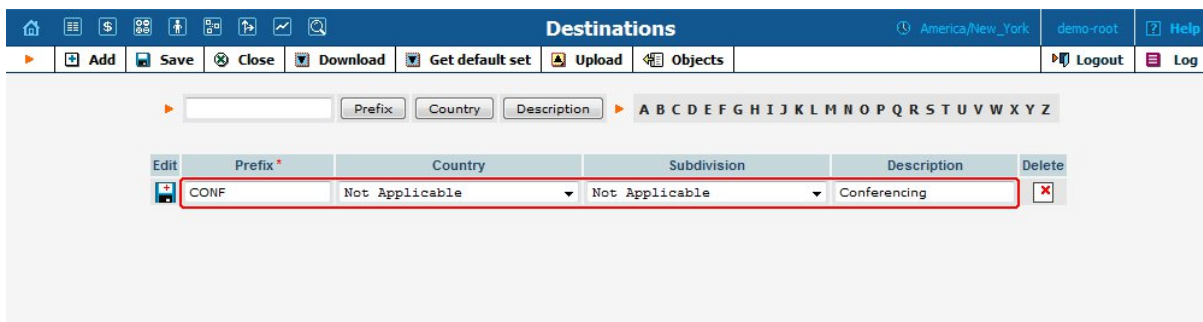


Figure 6. Adding destination CONF

Vendor Configuration

This section describes an easy way to configure a tariff associated with an internal vendor for conferencing service.

3. Vendor tariff configuration

Use the following parameters to add a vendor tariff:

- Name: TeliCon-t
- Applied to: Vendor
- Currency: select a currency
- Service: Conferencing.

Figure 7. Adding a vendor tariff

Add a rate for the destination **CONF** to the tariff.

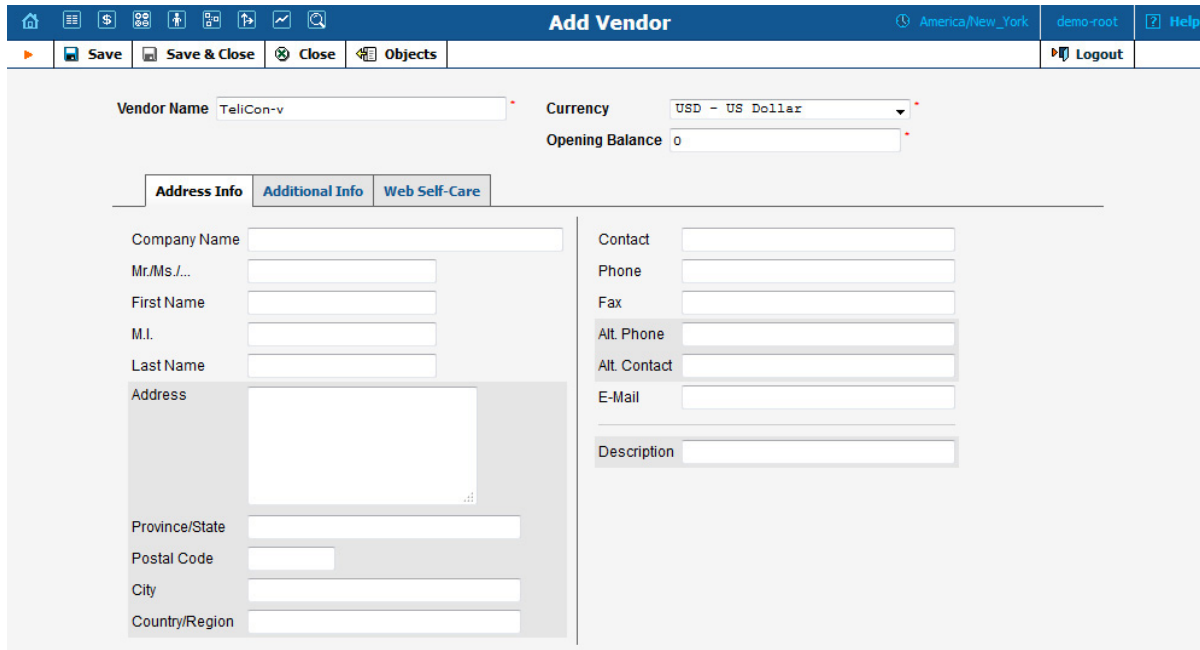
Edit	Destination *	Country	Type	Interval, seconds * >>	Price, USD/minute * >>	Effective From	
		Description				MM-DD-YYYY HH24:MI:SS *	
	CONF		NOT DEFINED	1	0	immediately	

Figure 8. Adding CONF to the vendor tariff

4. Adding a vendor for conferencing service

Create a new vendor as follows:

- Name: TeliCon-v
- Currency: The same as for the vendor tariff
- Opening Balance: 0.



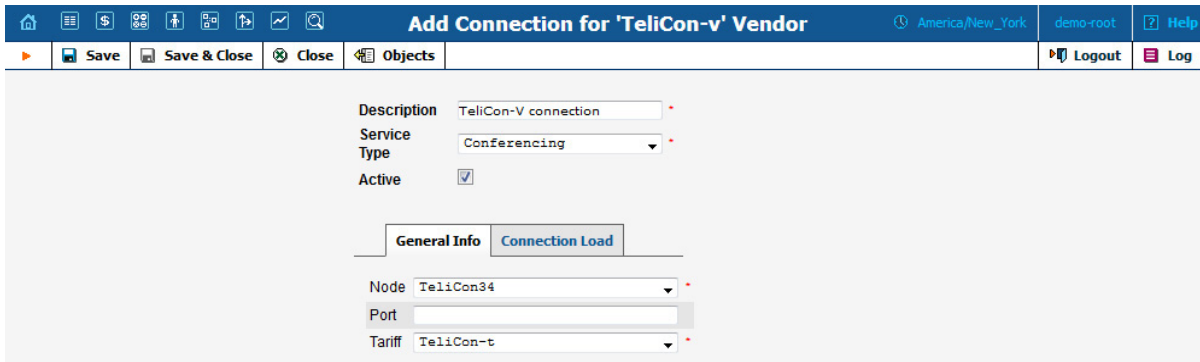
The screenshot shows a web application interface for adding a vendor. At the top, there is a blue header bar with the title "Add Vendor" and user information "America/New_York" and "demo-root". Below the header is a navigation bar with buttons for "Save", "Save & Close", "Close", and "Objects". The main form area has three tabs: "Address Info", "Additional Info", and "Web Self-Care". The "Address Info" tab is selected. The form contains the following fields:

- Vendor Name: TeliCon-v
- Currency: USD - US Dollar
- Opening Balance: 0
- Company Name
- Mr./Ms./J...
- First Name
- M.I.
- Last Name
- Address
- Province/State
- Postal Code
- City
- Country/Region
- Contact
- Phone
- Fax
- Alt. Phone
- Alt. Contact
- E-Mail
- Description

Figure 9. Adding a conferencing vendor

Add a new vendor connection:

- Name: TeliCon-V connection
- Service Type: Conferencing
- Node: TeliCon34 (the node with the number of your virtual partition)
- Port: not defined
- Tariff: TeliCon-t
- Capacity: 100.



The screenshot shows a web interface for adding a vendor connection. The title bar reads "Add Connection for 'TeliCon-v' Vendor". The interface includes a navigation menu with "Save", "Save & Close", "Close", and "Objects". The main content area has the following fields:

- Description: TeliCon-V connection
- Service Type: Conferencing
- Active:
- Node: TeliCon34
- Port: (empty)
- Tariff: TeliCon-t

There are two tabs: "General Info" (selected) and "Connection Load".

Figure 10. Adding a connection for the vendor

Account Provisioning

This section describes steps on setting up an access number for conferencing service. Before adding a conference room, an administrator needs to set up a product for the access number account, add tariffs for the combination of Conferencing and Voice Calls services, and edit the Service and Rating entry settings.

5. Adding a customer tariff for conferencing service

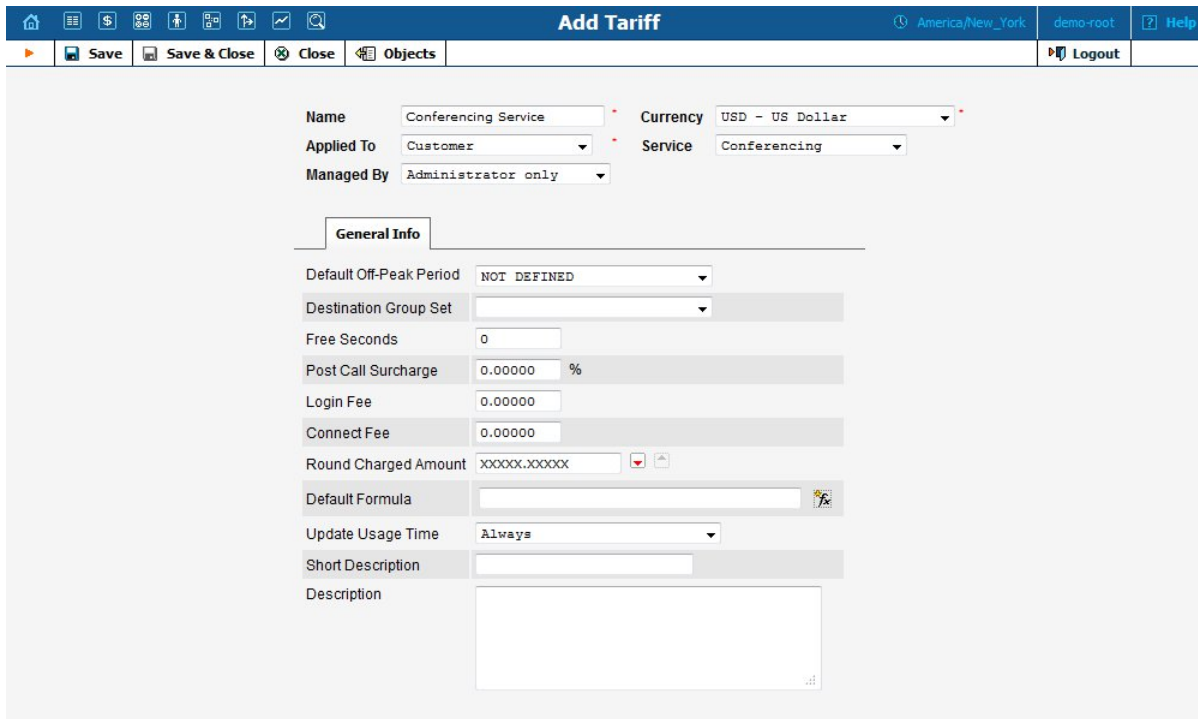
Add a tariff **Conferencing Service**, and configure it in the way described below. This tariff will be used to charge the access number account:

- Name: Conferencing Service
- Currency: select the currency
- Applied to: Customer
- Managed By: Administrator Only (default). Select a reseller to assign the tariff for use by a reseller in the system.
- Service: Conferencing.



The screenshot shows the 'Tariff Management' interface. At the top, there is a navigation bar with icons for home, list, currency, settings, user, and search. The title 'Tariff Management' is centered, with 'America/New_York' and 'demo-root' on the right. Below the navigation bar is a toolbar with buttons for 'Add', 'Close', 'Off-Peak Periods', 'xDR Re-rating', 'LCR Rates', and 'Objects'. A 'Logout' button is on the far right. The main area contains a search form with three dropdown menus labeled 'Applied To', 'Service', and 'Managed By', each with 'ANY' selected. A 'Search' button is to the right of these dropdowns. Below the search form is a table header with columns: Rates, Upload, Name, Currency, Applied To, Service, Managed By, Routing, Description, and Delete.

Figure 11. Adding a customer tariff



Add Tariff

Name: Conferencing Service * Currency: USD - US Dollar *
 Applied To: Customer * Service: Conferencing *
 Managed By: Administrator only *

General Info

Default Off-Peak Period: NOT DEFINED
 Destination Group Set:
 Free Seconds: 0
 Post Call Surcharge: 0.00000 %
 Login Fee: 0.00000
 Connect Fee: 0.00000
 Round Charged Amount: XXXXX.XXXXX
 Default Formula:
 Update Usage Time: Always
 Short Description:
 Description:

Figure 12. Conferencing tariff

Add a rate for the destination **CONF** to the tariff **Conferencing Service**.

6. Adding a customer tariff for voice calls service

Add a new tariff for **Voice Calls** service and configure it as follows; alternatively, you can use an existing tariff. The tariff will be used to charge calls made by a conference participants to the Conferencing server:

- Name: Voice-t
- Currency: select the currency
- Applied to: Customer
- Managed By: Administrator Only (default). Select a reseller to assign the tariff for use by a reseller in the system.
- Service: Voice Calls.

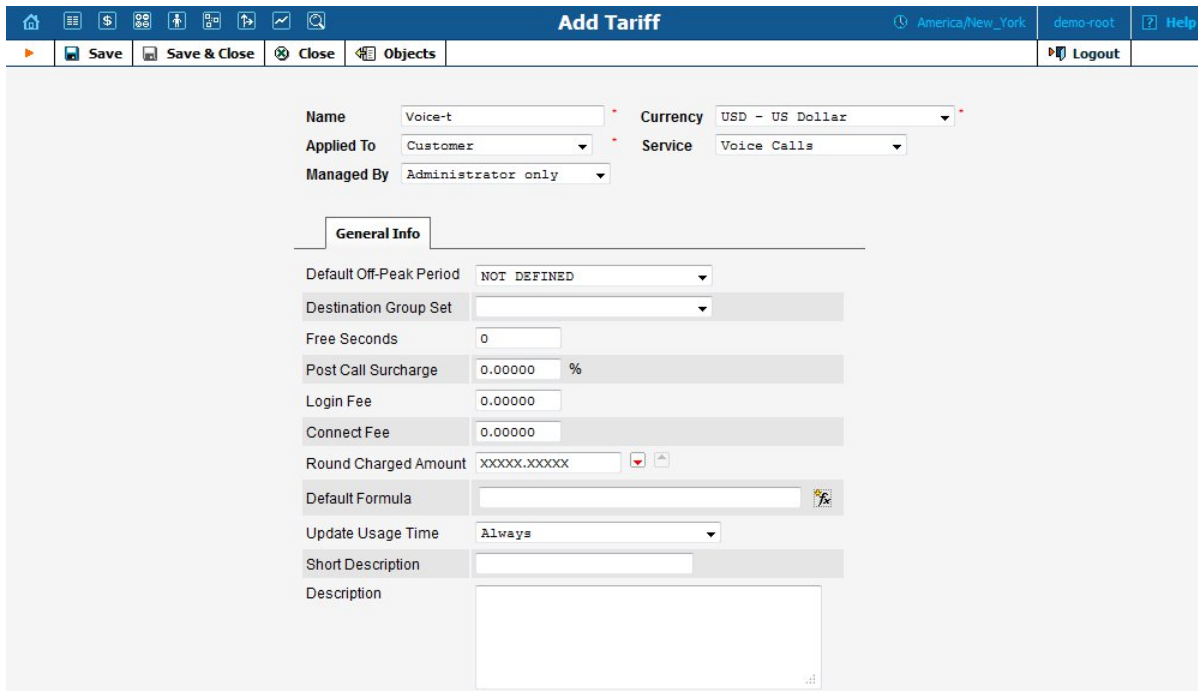


Figure 13. Voice Calls tariff

Add a rate for the destination **CONF** to the tariff **Voice-t**.

NOTE

If you are configuring the service for a reseller, add two additional tariffs for **Voice Calls** and **Conferencing** services applied to **Reseller**. Rates for the destination **CONF** must be present in these tariffs.

7. Product configuration

A product is a combination of services provided to a customer for a price. A product includes both **Voice Calls** and **Conferencing** types of service that an access number uses to forward calls to a conference room.

Add a new product called **Conference-p** for the conferencing service.

Product Name: *

Product Name visible to End User: *

Product Type: Main Product Add-on Product

Currency: *

Managed By:

Included Services | Service Configuration | Usage Charges | Volume Discount

Service type: Voice Calls

System built-in services: Voice Calls

Figure 14. Adding a new product

Rating entry is a component of a product. It specifies whether customers can use a service and how they should be charged for it. In the product’s settings add two **Usage Charges** entries using the tariffs created before.

The **Conferencing** entry:

- Service: Conferencing
- Node: TeliCon34
- Tariff: Conference-t
- Access Code: not defined.

Included Services		Service Configuration	Usage Charges	Volume Discount	Recurring Charges	Additional info	Notepad
Edit	Log	Service Used At					
		Node	Access Code	Service	Tariff		
Service type: Voice Calls							
		Any node	Any code	Voice Calls		Voice-t	
Service type: Conferencing							
		TeliCon34	Any code	Conferencing		Conferencing Service	

Figure 15. Rating entry for conferencing service

You can define the default value of service attributes for all accounts which the product will be assigned to. Select **Enabled** from the list to activate Conferencing service. The settings may be overridden on the account level.

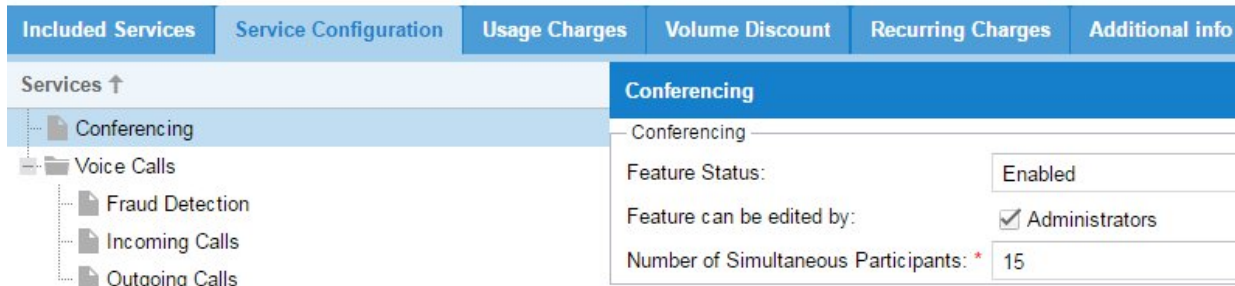


Figure 16. Feature status for conferencing service

The **Voice Calls** rating entry:

- Service: Voice Calls
- Node: ANY
- Tariff: Voice-t
- Access Code: not defined

Services	Service Configuration	Usage Charges	Volume Discount	Recurring Charges	Additional info	Notepad
Service Used At						
	Node	Access Code	Service	Tariff		
Voice Calls						
	Any node	Any code	Voice Calls	Voice-t		

Figure 17. Rating entry for voice calls service

8. Customer provisioning

Create a new Retail customer.

Make sure that the selected currency corresponds to the currency of the added product and customer tariff.

Add Customer

& Close
Close
Objects

Customer ID *

Blocked

Customer Class *

Balance Control

Currency *

Permanent Credit Limit

Address Info
Web Self-Care
Additional Info
Life Cycle
Payment Info
Volume Discounts
Service Configuration

Company Name

Mr./Ms./...

First Name

M.I.

Last Name

Address

Contact

Phone

Fax

Alt. Phone

Alt. Contact

E-Mail

BCC

Figure 18. Customer provisioning

9. Adding access number account

Add an account that will be used to access the conference. Telinta supports both prepaid (debit) and postpaid (credit) accounts. However, since Conferencing is a postpaid service, select credit type for the new account:

- Type: Credit
- Service Password: Auto

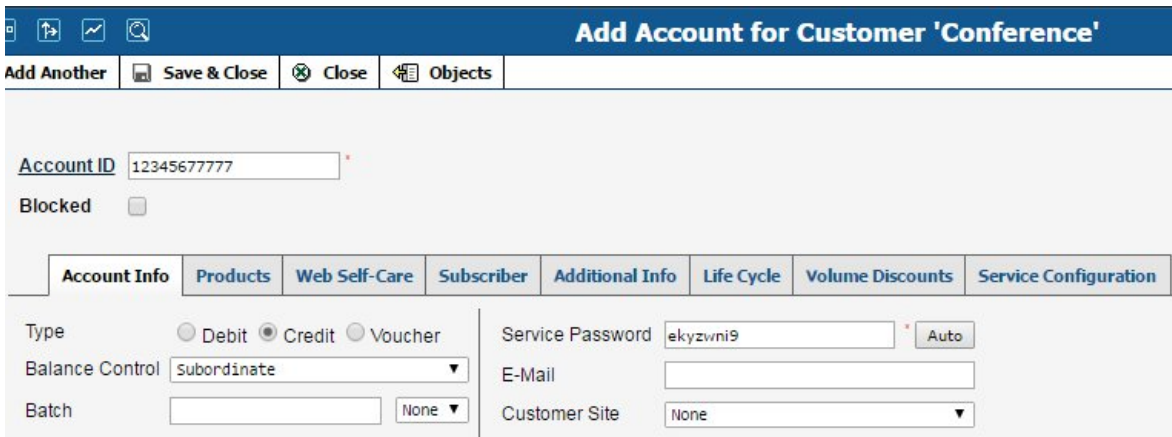


Figure 19. Adding an access number account

Choose the product that was previously added. Confirm the changes by selecting **Save** or **Save & Close**.

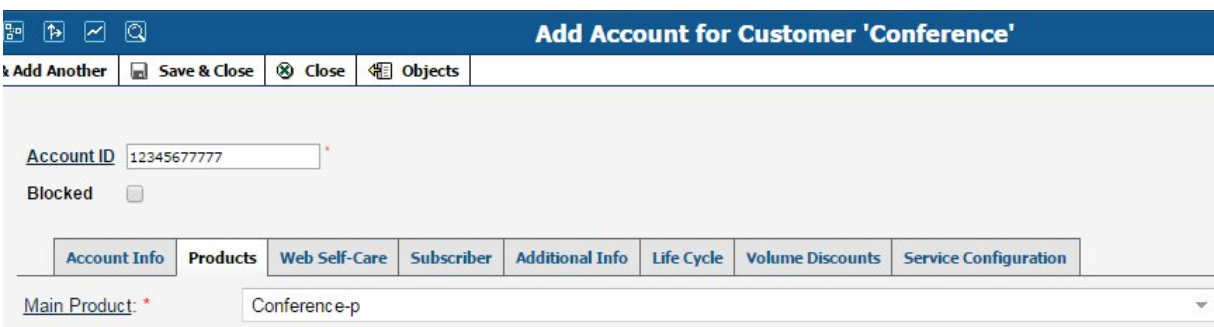


Figure 20. Selecting a product for the account

Set the Forward Mode to **Forward to SIP URI** under **Service Configuration** tab, **Incoming Calls** section. It is recommended to set **Default Answering Mode** to **Forward Only** for the access number account.

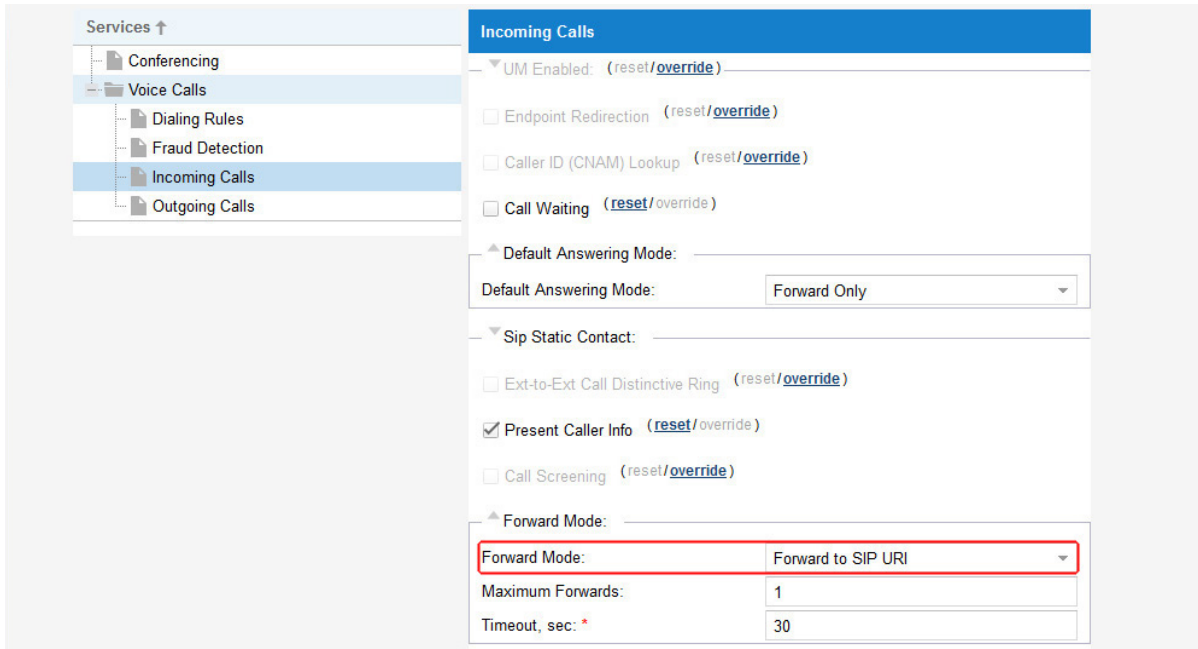


Figure 21. Incoming Calls section

Proceed to **Forward** TAB and add a SIP URI **CONF@199.180.223.108**, where **CONF** is a special destination which should be already present in the destinations list and **199.180.223.108** is the IP address of the conference server.

NOTE

A SIP URI vendor connection must exist in your SIP environment. If the connection doesn't exist or SIP option is disabled on the connection, calls to the conferencing server will fail.

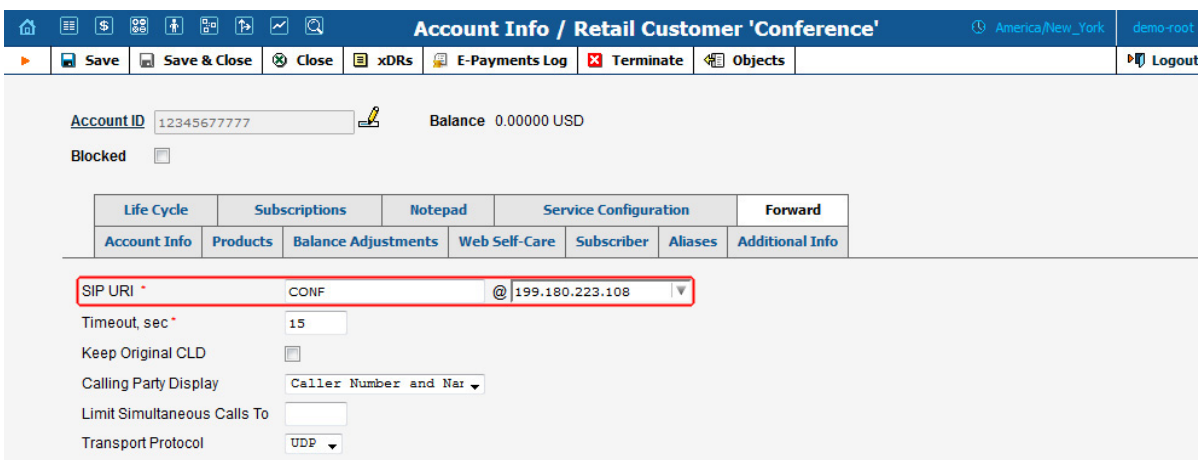


Figure 22. SIP URI

Open the tab **Service Configuration** on the Account Info page. Select **Conferencing** Service Type, select **Enabled** from **Conferencing** drop-down list, and define the number of participants.

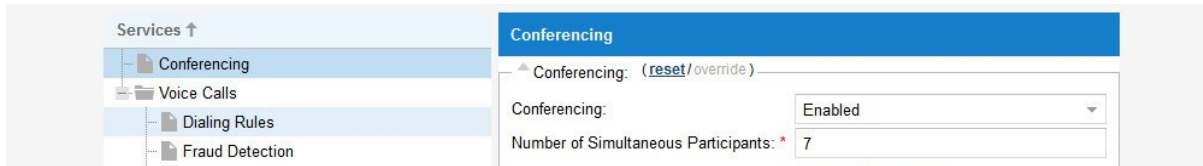


Figure 23. Enabling conferencing service

Account Self-Care Instructions

Conference rooms can be created at the Account Self-Care interface.

Log into the Account Self-care interface. Select **Conferences** tab under **IP Centrex** section at the left side of the page.

Click **Add** link to create a new conference room.

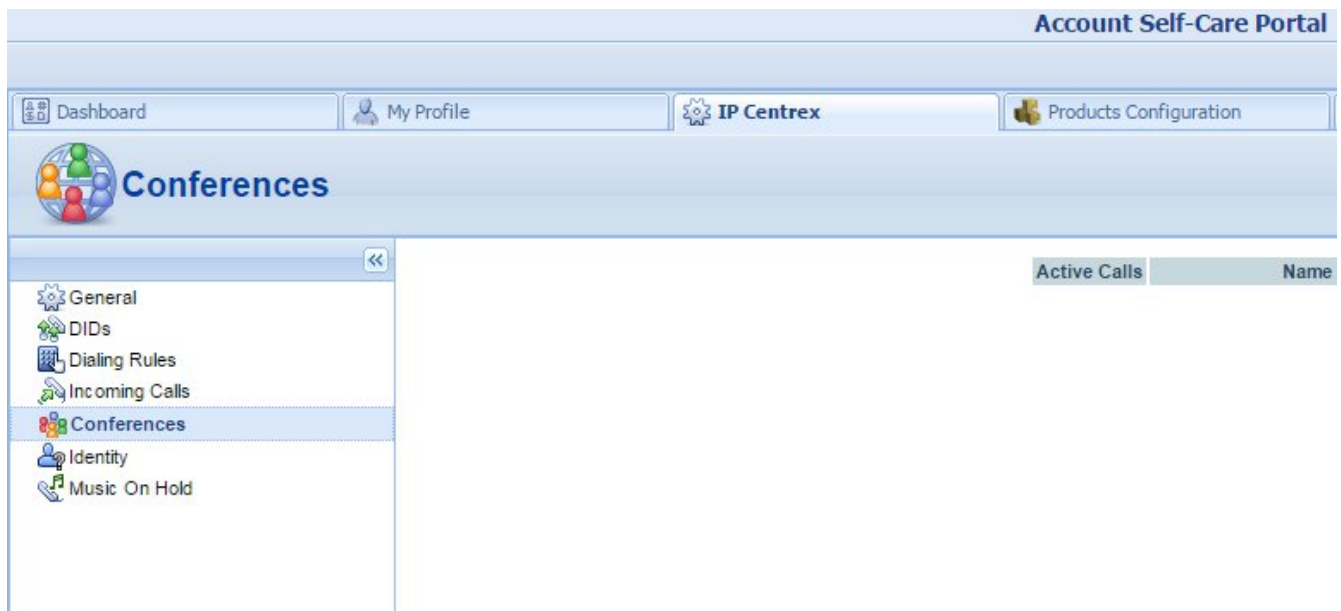
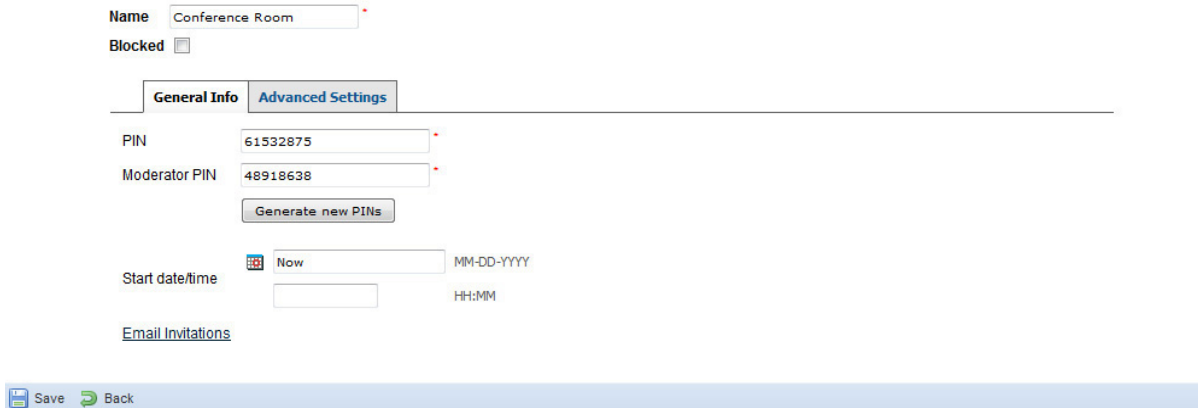


Figure 24. Conferences menu

10. Adding a conference room

Fill in the form under **General Info** tab. See the information below for a description of the available options.



The screenshot shows the 'General Info' tab of a conference room configuration page. At the top, there is a 'Name' field with the value 'Conference Room' and a red asterisk indicating it is mandatory. Below it is a 'Blocked' checkbox. Two tabs are visible: 'General Info' (selected) and 'Advanced Settings'. Under 'General Info', there are two mandatory fields: 'PIN' (61532875) and 'Moderator PIN' (48918638), both with red asterisks. A 'Generate new PINs' button is located below the Moderator PIN field. The 'Start date/time' field has a calendar icon, the text 'Now', and a 'MM-DD-YYYY' format indicator. Below it is a field for 'HH:MM'. At the bottom of the form is a link for 'Email Invitations'. A blue bar at the very bottom contains 'Save' and 'Back' buttons.

Figure 25. General Info tab

General Info tab has the following fields:

Name: A text description of the conference room.

Blocked: If activated, conference participants will not be able to access the conference room.

PIN: The code to enter the conference room.

Moderator PIN: The code used by the conference Moderator.

Generate new PINs: Click this button to generate a new set of codes. Any numeric PIN number is acceptable and can be used to access the conference. The PIN numbers should be unique.

Start date/time: Specifies the date and time when the conference will start. By default, a room can be used right after its creation (“Now”). Otherwise, **Start date/time** should be set by clicking the icon (📅) and selecting an appropriate date/time.

Email Invitations: Send an invitation email to the conference participants. Click this link to open default email client software on your computer for sending an email.

Mandatory fields are marked by *.

The **Advanced Settings** tab allows to configure the conference room options.

Name

Blocked

Wait for moderator:
 Lock the conference after all participants exit:
 Announce number of participants:

Limit conference duration:
 Conference duration: minutes
 Maximum participants:
 Mute options:
 Announce name:
 Conference recording:

Figure 26. Advanced Settings tab

The following options are available under **Advanced Settings** tab:

Wait for Moderator: If the option is enabled, conference participants will not be allowed to enter the room until the Moderator arrives.

Locking the room: When the last participant leaves the conference, the room is blocked automatically. Secure locking is implemented to prevent unauthorized use. You can opt to disable this feature and unblock the conference room by un-checking the “Blocked” box.

Announce number of participants: At the time of joining the conference, the IVR announces the number of participants in the room.

Conference recording: Activates the conversation recording. Call recording can be enabled automatically (**Auto** mode) or manually by the conference moderator (**On request** mode). The recording is available for download in mp3 format during 14 days.

Limit conference duration: It specifies the maximum time that the conference is active. A recording will be played to notify all participants 5 minutes before the end of a session.

Maximum participants: Specify the maximum number of participants who can enter the room. The **Moderator** can enter the room anytime, even if the limit was reached.

Mute options:

Participants: All participants can mute/unmute themselves via the touchtone commands.

Non-Moderators: You can select from two options for non-moderator muting:

- Only the Moderator is unmuted; other participants are muted by default, but can easily unmute themselves.
- Only the Moderator can speak during the conference. The rest of participants cannot unmute themselves (i.e., "Lecturing mode").

Announce name options: Announces the name of a new participant who enters the room.

No name announcement: A new participant enters the room without any announcement.

Announce name with review: Each participant is asked to record his or her name before entering the conference. The recording is played to the person before he or she enters the room. The conference participants are informed about the new person who enters the room and the recorded name is played.

Announce name without review: Each participant is prompted to record his or her name before entering the conference. When a new participant enters the room, the recording is played to the conference participants.

Music On Hold tab allows to configure the behavior of a music track playing within your conference. The tab is displayed only for already added rooms.

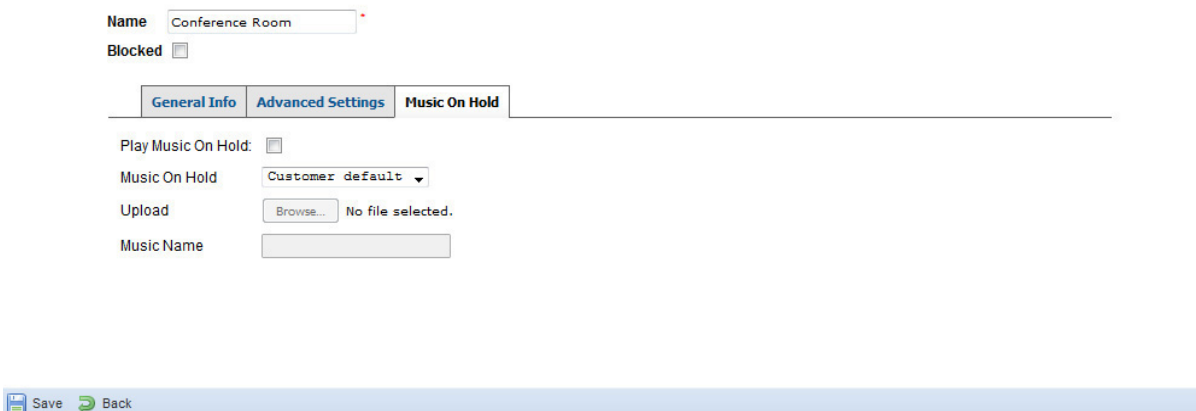


Figure 27. Music On Hold tab

Telinta provides music on hold for your conference. You can upload your own.

The fields available under **Music On Hold** tab include:

Play Music On Hold: Enables “Music On Hold” functionality for the room.

Music On Hold: Allows selecting a pre-defined MOH file from the drop-down list.

Upload: Allows uploading of a new MOH file to use for “Music On Hold”.

Music Name: Defines a custom name for the uploaded file.

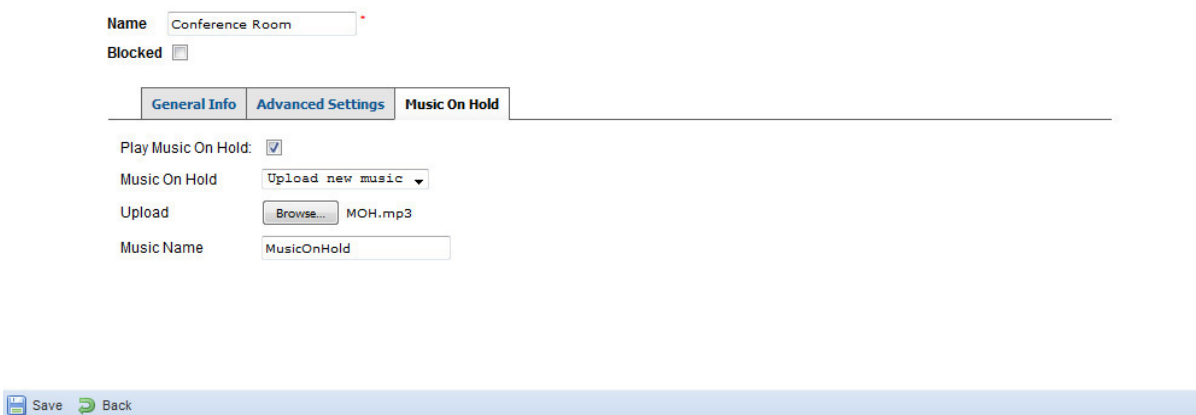


Figure 28. Uploading Music On Hold

To upload a new file, select **Upload new music** from the drop-down list **Music On Hold**.

IMPORTANT

Due to synchronization of servers, the procedure can take up to 10 minutes from uploading to appearing on the interface.

Accessing the Conference

Once all configuration tasks described above are completed, you can call the access number, and start using the service. Access numbers can be dialed from any phone. An account that has conference room settings can be used as an access number for reaching the conference, but it is possible to use separate accounts for access numbers.

To start using the conference room, follow these simple steps:

- Dial an access number.
- After the voice prompt, enter the PIN, and the system will recognize which room you want to enter.
- If the entered PIN was correct and you have been authorized to access the room, you will hear MOH and will be connected to the conference.

By default, the system requires to enter PIN number to access a conference room. It is possible to set up the conference to be accessed without entering PIN.

A conference can be scheduled for a specific time, and an invitation can be sent to its participants via email informing about its start date and time.

The system provides access to the conference room settings via the web interface and SOAP/REST API.

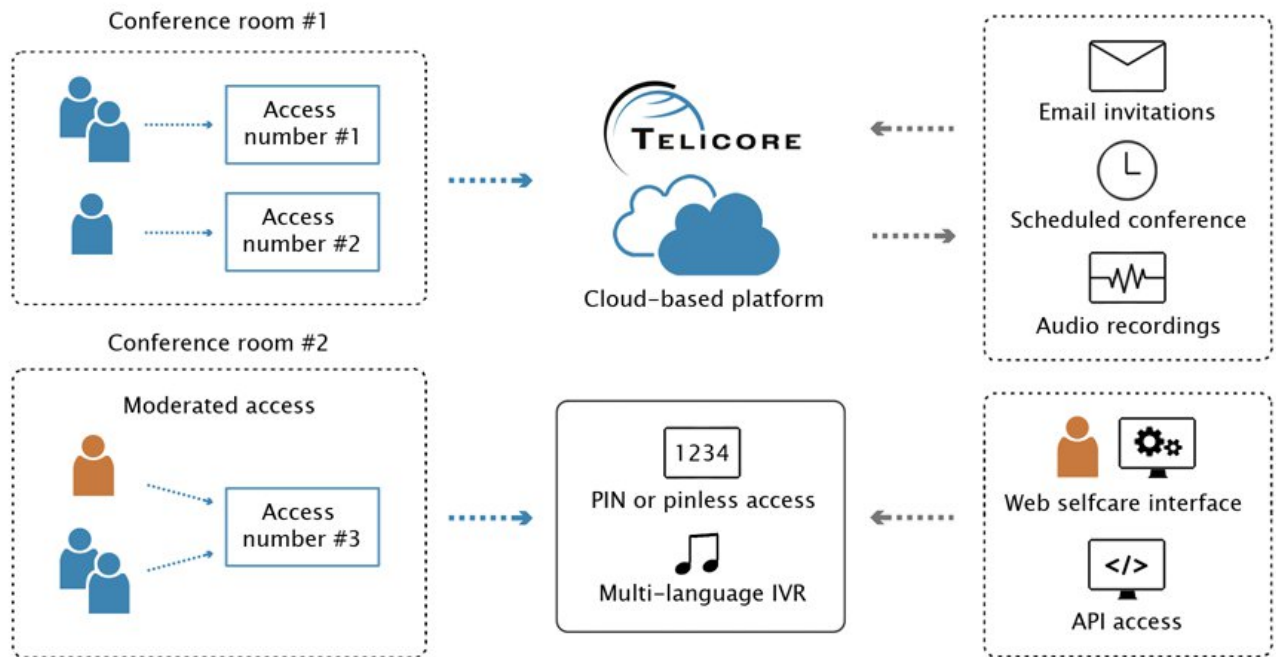


Figure 29. Overview of conferencing service

TeliCon Features

Conference Call Recording

TeliCon allows recording any or all conference calls. This feature is available for the Moderators only. Conference call recording can be started by pressing *** *** on the Moderator’s touch-tone keypad during the conference call. The recording will stop automatically when the conference call is finished.

NOTE

If the **Auto** recording mode is activated, all calls will be recorded automatically. The recording will be activated only if there is at least one participant in the conference room.

To download a recording, open Account self-care and follow to the **Conferences** tab under **IP Centrex** section.

Active Calls	Name	Recordings	Status	Delete
	Conference Room			
	Room 4			

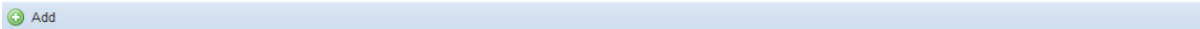







Figure 30. Accessing the conference list

Click on the **Recordings** icon next to the respective room to download audio recordings.

Active Calls	Name	Recordings	Status	Delete
	Conference Room			
	Room 4			


 Add

Figure 31. Link to download

The Recordings page provides a list of audio records ordered by call date and time that are available for download by clicking the corresponding hyperlinks. Each recording is available in mp3 and wav format.


Name	Size, Mb	Duration, Mmm:sec	Delete
2016-01-01 09:07:39 (download original wav)	3.2	05:21	

Figure 32. Recordings page

Recordings are available for download for 14 days following the call.

Key Codes

The following commands are available to control conference calls from the touch-tone keypad:

*** 1** Self-Mute

To mute your line, enter *** 1**. To unmute your line, enter *** 1** again.

*** 2** Lock/Unlock conference

This feature allows the Moderator to lock the conference in order that no participants can join it. To lock the conference, the Moderator enters *** 2**. To unlock the conference, the Moderator enters *** 2** again.

To request unmuting, a participant presses *** 2**, and the Moderator can see this request on the web-interface.

*** 3** Disconnect last person who joined the conference.

*** 4** Increase the listening volume.

*** 5** Extend the conference by 15 minutes.

*** 6** Decrease the listening volume.

*** 7** Decrease the speaking volume.

*** 8** Exit the menu.

*** 9** Increase the speaking volume.

*** *** Start recording (Moderator only). Recording will stop automatically when the conference ends.

Customizable CODECs

By default, the G.711u codec is used for audio streams. However, Telinta can enable additional CODECs, such as G.711a or GSM upon request.


Active Calls

The **Active Calls** screen contains information about the conference participants and provides the functionality to manage them.

Page refresh interval: 60 sec.

CLI	Status	Requests	Kick
020020 (020020) (Admin)	Unmuted	-	
020123456 (020123456)	Unmuted	-	
Mute all			

Figure 33. Active Conference Calls for the room

The Moderator can drop any participant by clicking **Kick** icon () on the **Active Calls** screen.

To mute or unmute a participant, click on **Status** link.

Page refresh interval: 60 sec.

CLI	Status	Requests	Kick
020020 (020020) (Admin)	Unmuted	-	
020123456 (020123456)	Unmuted	-	
Mute all			

Figure 34. Muting and unmuting participants

To change the default participant's ID to some alias, click **CLI** link.

Page refresh interval: 60 sec.

CLI	Status	Requests	Kick
020020 (020020) (Admin)	Unmuted	-	
020123456 (020123456)	Unmuted	-	
Mute all			

Figure 35. Participant's ID

Then define a name which should be assigned to this participant.

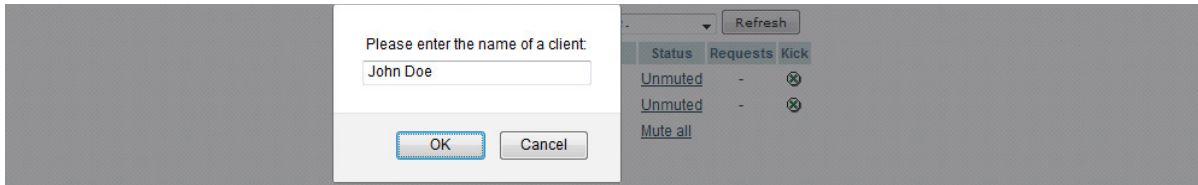


Figure 36. Adding an alias

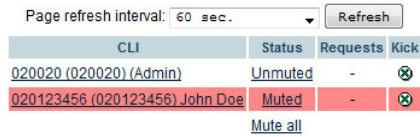


Figure 37. Muted participants

When a muted participant presses *** 2** the Moderator will see this request on the corresponding screen and can act accordingly.

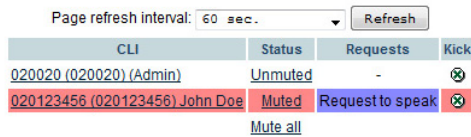


Figure 38. Request to speak from a muted participant

Languages

Audio prompts in the following languages are currently available:

- English
- French (fr)
- German (de)
- Italian (it)
- Polish (pl)
- Russian (ru)
- Spanish (es)

By default, English language is used. To use a different language for your conference, you should set special SIP URI settings for the account that forwards call to the conference server.

To use the special settings set Forward Mode to **Forward to SIP URI** under the tab **Service Configuration, Incoming Calls** section of the account. Under the **Forward** TAB, set a SIP URI **CONFL<language code>@199.180.223.108**, where **CONF** (optionally followed by a number) is a special destination, **L** is the required parameter, **<language code>** is a two-letter language code, and **199.180.223.108** is the IP address of the conference server.

For example, to select Spanish language, enter the SIP URI **CONFLes@199.180.223.108**

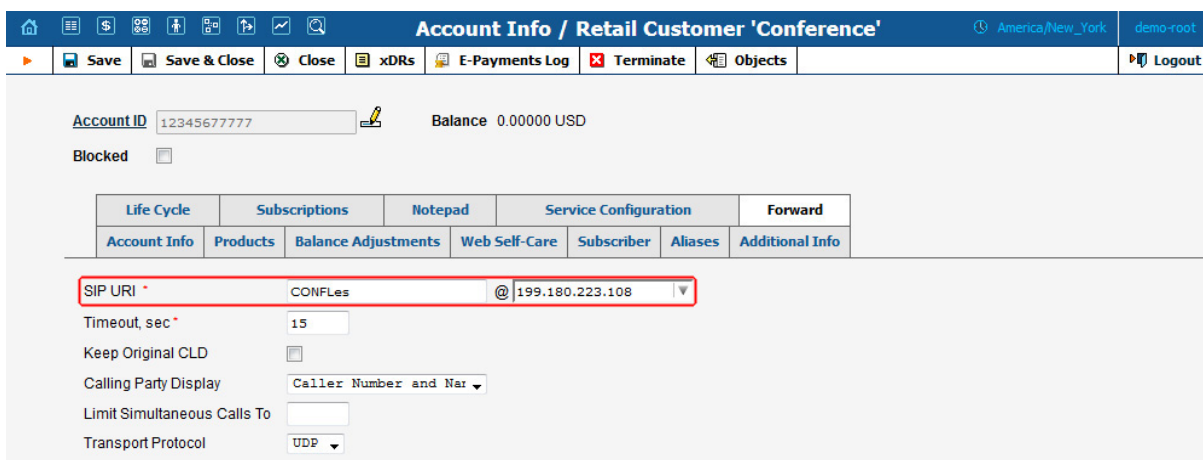


Figure 39. Selecting language for the conference

The system allows uploading branded audio recordings for your conference. If you require an additional language not shown on the above list, or would like to upload your own privately branded recordings, you can upload your own recorded prompts.

Please contact support@telinta.com for more information on installing customized audio prompts.

Pinless Access

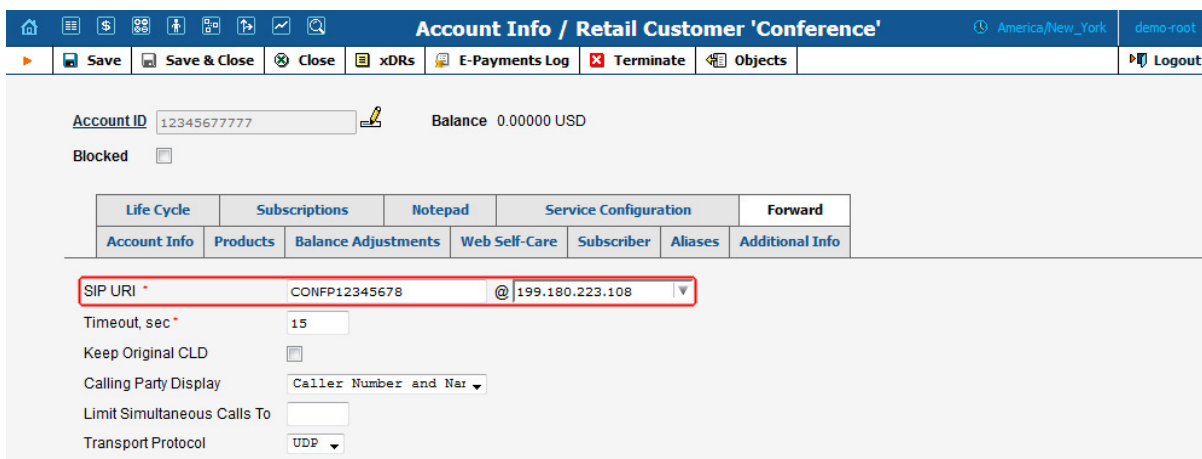
TeliCon provides users the ability to access a conference room without entering a PIN number. Instead of asking a caller to enter a conference PIN number, the system can automatically use PIN of a pre-configured conference room.

To configure this option, add a SIP URI redirect with a special parameter for an account you call to access the conference.

Use the pattern **CONF<pin number>@199.180.223.108**

Where **CONF** (optionally followed by a number) is a special destination, **P** is the required parameter, **<pin number>** is PIN or Moderator PIN of a conference room, and **199.180.223.108** is the IP address of the conference server.

For example, to use the PIN number 12345678 for the conference, use the SIP URI **CONF12345678@199.180.223.108**.



The screenshot shows a web interface for configuring an account. The title bar reads "Account Info / Retail Customer 'Conference'". Below the title bar are navigation buttons: Save, Save & Close, Close, xDRs, E-Payments Log, Terminate, Objects, and Logout. The main content area shows "Account ID" as 1234567777 and "Balance" as 0.00000 USD. There is a "Blocked" checkbox. Below this is a table with tabs: Life Cycle, Subscriptions, Notepad, Service Configuration, and Forward. Under "Service Configuration", there are sub-tabs: Account Info, Products, Balance Adjustments, Web Self-Care, Subscriber, Aliases, and Additional Info. The "SIP URI" field is highlighted with a red box and contains the text "CONF12345678@199.180.223.108". Other fields include "Timeout, sec" (15), "Keep Original CLD" (checkbox), "Calling Party Display" (Caller Number and Nat), "Limit Simultaneous Calls To", and "Transport Protocol" (UDP).

Figure 40. Pinless access

NOTE

Make sure the option **Wait for moderator** is disabled in the settings of the conference room if you decide to use Moderator PIN number for the conference participants.

Troubleshooting Your Conference

Symptom	Problem	Action
Calls to the conference fail.	The conference server rejects calls.	<p>Do one of the following steps:</p> <ul style="list-style-type: none"> • Check that a SIP-URI connection exists in the environment and is not disabled. • Make sure a vendor with the Conferencing service exists in the environment. • Check configuration of the product assigned to the account that forwards calls to the conference. Make sure that the product has rating entries for both Voice Calls and Conferencing services. • Check tariff rates. Both tariff for Voice Calls and Conferencing services must have rates for CONF destination.
You can't reach the conference from particular locations, such as PSTN or cell phones.	There are no compatible CODECs to connect the call with the conference server, in particular g711u (PCMU) and g711a (PCMA).	<p>Do one of the following steps:</p> <ul style="list-style-type: none"> • Check configuration of a calling soft- or hard phone or IP PBX and enable the CODECs. • If you are calling from a PSTN or cell phone, contact the DID provider to enable the CODECs.

Symptom	Problem	Action
The conference does not respond to dialing a PIN number.	DTMF tones are not properly transmitted.	Do one of the following steps: <ul style="list-style-type: none"> • Check configuration of a calling soft- or hard phone or IP PBX. • If you are calling from a PSTN or cell phone, contact the DID provider and ask to investigate the problem on their side.
The conference does not accept a PIN or moderator PIN numbers which are configured for a conference room.	The product assigned to the account that forwards calls to the conference is missing a rating entry for the service type Conferencing .	Add a rating entry for Conferencing service to the product used by the access number account (<i>Figure 15. Rating entry for conferencing service</i>).
	The conference room has start date/time in future.	Change date/time settings of the conference room.
	The option Close after all participants exit had been enabled and the conference room was blocked.	Unblock the conference room.
The conference is locked.	The Limit conference duration option was enabled for the room and the conference has been outdated.	Change configuration of the conference room.

Definitions

The following terms have specific meanings in this document:

Conference Call: A telephone call in which the calling party joins more than one participant on the same call. Conference calls allow the called party to participate during the call, or the call may be set up so that the called party only listens in and cannot be heard by other participants.

Conference Room: A virtual place where all participants can talk to each other. The room configuration is flexible, and can be configured to provide/restrict various features for the participants.

Moderator: The person leading the conference. The moderator is granted access to perform variety of administrative actions during the conference call (e.g. muting/unmuting and dropping of participants).

Participant: A person who participates in the conference call.

Recording: A digital recording of the conference. A recording is available for immediate download after the conference is finished. Recorded media files remain available for 14 days.

Reservationless Conference: A permanently enabled conference that allows conducting a conference call at any time without having to make a prior reservation.

Scheduled Conference: A conference that allows conducting a call only at a specified time defined in the room settings.