

**Edition 1, SW Release 3.1.8 and higher, December 2006**

# Table of Contents

## Manual-I: see Installation Guide

This manual described step-by-step guide to install and configure QuadroCS basically.

## Manual-II: see Administrator's Guide

This manual describes in detail the menus available to administrator only.

## Manual-III: Extension User's Guide

<b>About this Moderator's Guide .....</b>	<b>3</b>
<b>QuadroCS' Graphical Interface .....</b>	<b>4</b>
<b>Recurrent Buttons .....</b>	<b>5</b>
<b>Recurrent Functional Buttons of the GUI .....</b>	<b>5</b>
<b>Entering a SIP Addresses correctly .....</b>	<b>5</b>
<b>Moderator's Menus .....</b>	<b>6</b>
<b>Conference Moderator's Main Page .....</b>	<b>6</b>
<b>Conference Menu.....</b>	<b>7</b>
Conference Progress .....	7
Activate Conference .....	9
Send Notification Mail.....	9
Recorded Conferences.....	9
Conference Statistics.....	10
<b>Conference Properties Menu.....</b>	<b>12</b>
General Settings .....	12
SIP Registration Settings .....	13
SIP Advanced Settings .....	13
Recording Settings.....	14
Customization .....	14
Participants.....	15
New Participants Configuration .....	17
Handset Added Participants Configuration.....	17
Schedule .....	18
Reservation.....	19
Codecs .....	20
AGC Settings.....	21
<b>Logout .....</b>	<b>21</b>
<b>Feature Codes .....</b>	<b>22</b>
<b>QuadroCS' Auto Attendant Services.....</b>	<b>23</b>
<b>Appendix: Glossary.....</b>	<b>24</b>

## About this Moderator's Guide

The QuadroCS Manual is divided into three parts:

- **Manual-I: Installation Guide** gives step-by-step instructions to provision the Quadro Conference Server and configure the conferences with the Epygi SIP Server. After successfully configuring the Quadro, a user will be able to automatically or manually initiate the conferences between several IP peers, manipulate with the active conferences and record them if needed.
- **Manual-II: Administrator's Guide** explains all Quadro Conference Server management menus available for administrators only. Further, it includes a list of all System Default Values.
- **Manual-III: Moderator's Guide** explains all menus that can be accessed and configured by the conference users within the QuadroCS. Further, a list of all available conference codes can be found there.

In detail the Moderator's Guide includes the following chapters:

[QuadroCS' Graphical Interface](#) introduces to the Quadro's graphical user interface and explains all recurrent buttons.

[Conference Menu](#) explains the moderator specific input options used to manage and configure the conference.

[Feature Codes](#) lists all feature codes available during the conference.

[QuadroCS' Auto Attendant Services](#) introduces functions available in the QuadroCS' auto attendant.

[Appendix: Glossary](#) explains some technical terms.


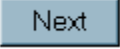

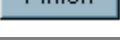

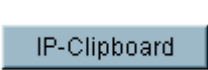
## QuadroCS' Graphical Interface







The QuadroCS configuration management may be accessed on three different levels: Administrators, Moderators and Participants. Moderators and Participants have to login with the conference ID or Username and the particular passwords specified by the Administrator for each conference. Moderators are able to modify the login password from the Conference Properties page. Participants have merely a read-only access to the Conference Server GUI.



Fig. III-1: Quadro Extension User's page

## Recurrent Buttons

Button	Description
	This button leads back to the previous page of a fixed sequence of pages (used mainly in wizards).
	This button leads forward to the next page of a fixed sequence of pages (used mainly in wizards).
	This button discards the latest not yet confirmed entries.
	This is the last button of a fixed sequence of pages that completes and saves the entries of an entire sequence.
	This button opens the help page belonging to the currently active Quadro management page.
	This button opens a window where the last inserted IP addresses are listed. It allows the user to make a quick selection of an IP address that has been previously used. This will avoid the user needing type it again. The clipboard can hold up to 10 IP addresses and a new IP address will replace the oldest one from the list.

Button	Description
	This button returns you to the page you were previously on.
	This button confirms an operation you started before.
	This button confirms an operation you chose before.
	This button discards an operation you chose before.
	This button saves the settings modified on the currently active management page.
	This button opens a window where the last inserted SIP addresses are listed. It allows the user to make a quick selection of an IP address that has been previously used. This will avoid the user needing type it again. The clipboard can hold up to 10 SIP addresses and a new SIP address will replace the oldest one from the list.

## Recurrent Functional Buttons of the GUI

In connection with tables, the following buttons - among others - usually occur:

Functional Button	Description
<b>Add</b>	Allows adding a new record to the displayed table. A new page will be displayed to enter any new settings.
<b>Edit</b>	Allows modifying the settings of the record selected by a checkbox. Normally only one (1) record may be selected. A new page will be displayed to enter the modified settings.
<b>Delete</b>	Deletes the selected entry(s) of a table. A warning message will ask for confirmation before deleting an existing entry.
<b>Select All</b>	Selects all table entry(s) for example for further deletion.
<b>Inverse Selection</b>	Inverses (opposites) an existing selection of table entry(s). If no entries are selected, clicking the button will select all records.
<b>Refresh in...</b>	May be shown in the upper right corner of a page. It displays the number of seconds remaining until the next refresh of the page will occur. It may be used to reload the page manually.

Most of the tables offer the option to sort the entries in ascending or descending order by clicking the headings of the columns. A small arrow next to the column heading indicates the direction of sorting - upward or downward. The entries of the table can be selected by using the corresponding checkboxes in order to edit or delete them.

## Entering a SIP Addresses correctly

Calls over IP are implemented based on Session Initiation Protocol (SIP) on the QuadroCS. When making a call to a destination that is somewhere on the Internet, SIP address must be given.

SIP addresses have to be specified in one of the following formats:

```
"display name" <username@ipaddress:port>
"display name" <username@ipaddress>
username@ipaddress:port
username@ipaddress
username
```

The display name and the port number are optional parameters in the SIP address. If the port is not specified, 5060 will be set up as the default one. The range of valid ports is between 1024 and 65536.

A flexible structure of wildcards is allowed. In comparison with a wildcard, the "?" character stands for only one unknown digit and the "\*" character stands for any number of any digits.

Particularly the following combinations can be used for your convenience:

- \*@ipaddress - any user from the specified SIP server
- username@\* - a specified user from any SIP server
- \*@\* - any user from any SIP server

**Please Note:** Wildcards are available for caller addresses only. No wildcard characters are allowed for called party addresses. To use "\*" and "?" themselves (as non wildcard characters), use "\\*" and "\?" correspondingly

# Moderator's Menus

## Conference Moderator's Main Page

After logging in as a moderator, the page **Conference Progress** is displayed. Here you may see the active conferences and the participants. From this page you may also access the settings of the conference to operate and perform actions that are available only to the moderator of each conference.

- **Conference Menu**
  - [Conference Progress](#)
  - [Activate Conference](#)
  - [Send Notification Mail](#)
  - [Recorded Conferences](#)
  - [Conference Statistics](#)
- **Conference Properties Menu**
  - [General Settings](#)
  - [SIP Registration Settings](#)
  - [Recording Settings](#)
  - [Customization](#)
  - [Participants](#)
  - [Schedule](#)
  - [Reservation](#)
  - [Codecs](#)
  - [AGC Settings](#)
- [Logout](#)

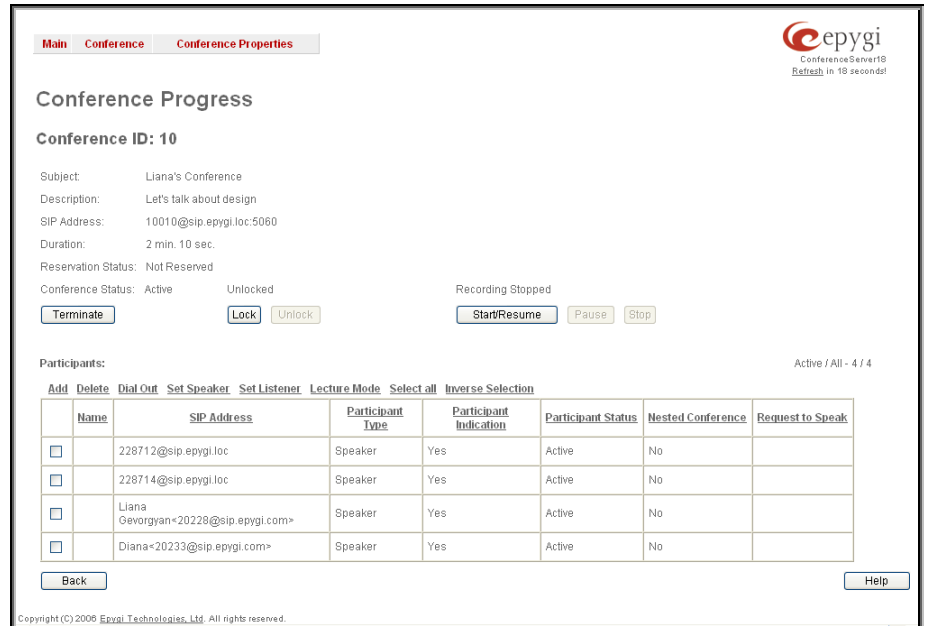


Fig. III-2: QuadroCS Moderator's page

## Conference Menu



Fig. III-3: Moderator Settings Dynamo Menu



Fig. III-4: Moderator's Settings Tree Menu

## Conference Progress

The **Conference Progress** page displays information about the conference, including the list of participants, and allows moderator to manage the conference.

The following read-only data is displayed on this page:

- Conference ID** – the unique ID on the conference.
- Info Text** – displays the text uploaded in the Info File from [Customization](#) page. In the picture illustration on the right side, the Info Text says "WELCOME to EPYGI's CONFERENCE!!!".
- Subject** - an optional subject for the conference.
- Description** - an optional description of the conference.
- SIP Address** - the SIP address of the corresponding conference.
- Duration** – the time the current conference is active.
- Reservation Status** – information whether this conference is a reserved one or not, also a count about maximum number of participant and an expiration time until the conference will be active.
- Conference Status** – the conference status (active, not active or waiting). In the conference is active, the information whether the conference is locked or not, and the recording status (recording started, recording paused and recording stopped) is also displayed herein.

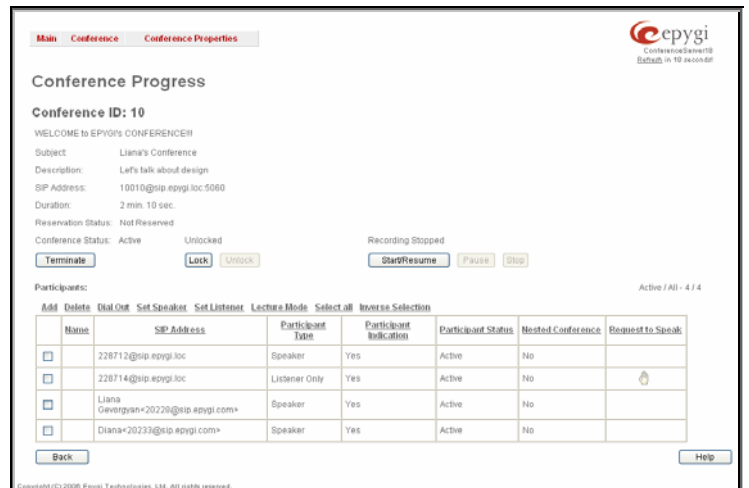


Fig. III-5: Conference Progress Page

The following buttons are available on this page to manage the active conference:

- Activate** – available for inactive conference only and used to activate a conference.
- Terminate** – available for active conference only and used to terminate the active conference
- Lock** – available for active conference only and used to lock the conference. When conference is locked, no users can connect it.
- Unlock** - available for active conference only and used to unlock the conference.
- Start/Resume** - available for active conference only and used to start the recording of the conference or to resume the recording if it was paused.
- Pause** - available for active conference only and used to pause the recording of the conference.
- Stop** - available for active conference only and used to stop the recording of the conference.

**Please Note:** **Pausing and Resuming** the conference recording can be used to edit the recorded conference audio file. When pause/resume operations are used, conference is recorded in a single file, leaving out the conversation during which conference recording was paused. When using stop/start operations, new files are created each time conference recording is started. All recorded conferences are listed in the [Recorded Conferences](#) page only after conference recording termination. In case of **pause/resume**, the recorded file is not terminated. In case of **stop/start** recording starts in new file.

The table of participants on this page lists all preconfigured participants (independent on the conference status), as well as new participants joined the conference (if still connected to the conference) and those participants added from the handset or GUI (unless the conference is terminated).

For active conference the table also displays participants added manually from GUI or from the handset and those participants that called in to the conference.

The **Conference Progress** table contains the following information for each participant.

**Name** – this information is specific to manually added participants only (see below).

**SIP Address** – indicates the SIP address of the participant.

**Participant Type** – indicates whether the participant is a speaker or a listener only.

**Dial Out** – this column is only present for inactive conferences and indicates whether or not it is a dial-out participant, i.e. whether the QuadroCS calls the participant when the conference is activated.

**Trigger** – this column is only present for inactive conferences and indicates whether the participant is a key member to start the conference, i.e. when participant dials in to the conference, the conference is getting automatically activated and the dial out participants (if any) are being called to join the conference.

**Participant Indication** – indicates whether or not a beep indication during the call conference is configured for this participant to be played when he joins or leaves the conference.

**Participant Status** – this column is only present for active conferences and indicates the state of the participant (active, for participants currently in the conference, not active, for participants not in the conference and joining, for participants currently joining but not yet connected to the conference).

**Nested Conference** – indicates if the participant acts as a nested conference or not.

**Duplication** – this column is only present for inactive conferences and indicates whether a participant with the same SIP address can join the conference.

**Request to Speak** - this column is only present for active conferences and indicates whether a listener participant has been requested to speak (by dialing \*9 from the handset, see Feature Codes). When a listener participant requests to speak, a hand-up icon appears in this column. Clicking on the hand icon in this column will grant the speaker permission to the corresponding participant. Participant with the speaker permissions are able to speak to the conference.

The following functional buttons are present on **Conference Progress** page to manipulate with the participants in the conference:

**Add** functional button opens an **Add Participant** page where a new participant can be manually added to the conference. The **Conference Progress – Add Participant** page consists of the following components:

**Participant Name** requires optional information (first name, last name, nickname, etc.) about the participant.

**SIP Address/Tel. number** requires the contact phone number (SIP address or Routing Number) of the participant. This number automatically will be dialed by the QuadroCS when the participant is configured to be a Dial Out (see below) or when a corresponding Conference Code is used (see Conference Codes).

The participant's SIP address should be a combination of username@hostaddress:port (where hostaddress can be an IP address, for example, 192.168.90.10, or a host name, e.g., sip.epygi.com). The port number is optional for the SIP address. If no port is specified, 5060 will be used. The range of valid ports is between 1024 and 65536.

The value will be implied as a Routing Number when either digits starting with zero (0) are inserted in this field or Routing Mode is enabled for the conference. In these cases, the Routing Number will be parsed through the Local Routing Table when placing a call to the corresponding participant. Otherwise, when Routing Mode is disabled, any numerical value starting with a non-zero digit will be treated as a SIP username and will be applied with the host name of the SIP server the conference is currently registered on.

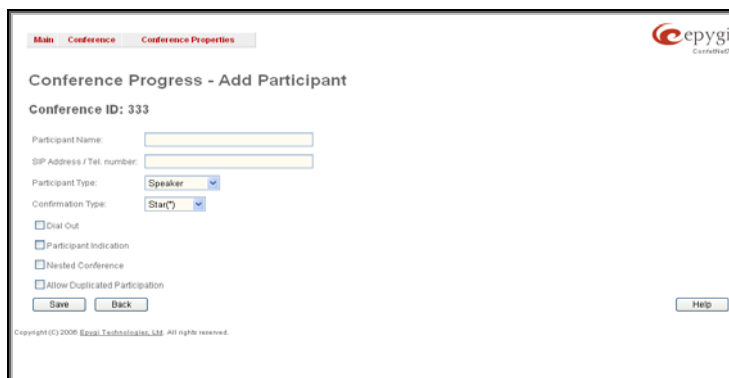


Fig. III-6: Conference Progress – Add Participant Page

**Please Note:** A direct call will be placed toward a participant's SIP address if the corresponding conference is registered on a different SIP server than the participant is registered on, or if the participant is not registered on the SIP server at all.

**Participant Type** list is used to select the type (speaker or listener) of participant in the corresponding conference.

**Confirmation Type** list is used to set the password protection for the participant joining the active conference. **Star (\*)** selection allows the participant to accept the conference invitation by pressing the \* button. Only participants connected to the conference with the moderator password will be provided with permissions to manipulate the conference.



A group of checkboxes on this page allow configuration of participant specific settings related to the corresponding conference:

- When the **Dial Out** checkbox is selected, the certain participant will be automatically dialed out when the conference is activated.
- **Participant Indication** enables the beep indication during the conference when this participant joins or leaves the conference.
- **Nested Conference** must be selected if the participant is a Conference itself and enables the correct behavior of conference termination.
- **Allow Duplicated Participation** checkbox allows multiple participants with the selected Caller ID (calling address) to join the corresponding conference. This is applicable when different participants are using the same shared number to place a call.

**Dial Out** functional button is used call to one or more inactive participant(s) inviting them to join the conference.

**Set Speaker** functional button is used to grant a speaker permission to one or more selected participant(s). Participant with the speaker permissions is able to speak to the conference.

**Set Listener** functional button is used to grant a listener permission to one or more selected participant(s). Participant with the listener permissions is not able to speak to the conference and are only listeners.

**Lecture Mode** functional button is used to grant a lecturer permission to a participant. Either listener or speaker participants can get the lecturer permission. Enabling lecture mode for a participant will allow him to speak to the conference and will mute all other participants of the conference.

**Please Note:** Only one participant can act in a lecture mode at the same time.

## Activate Conference

This link is used to activate a conference.

## Send Notification Mail

This link is used to send an email to the participants notifying them about the start of a conference and inviting them to join. The text of the notification email is being configured by the administrator.

## Recorded Conferences

Conference recording service allows you to record conferences and save them on the QuadroCS or the external USB Flash memory (depending on the configuration). To use conference recording service, it should be enabled from the [Recording Settings](#) page. The maximum duration of the recorded conference can be optionally limited from the [Recording Settings](#) page.

Conference recording can be manipulated either from the [Conference Progress](#) page or from the handset (see Feature Codes). If the **Recording Indication** is also enabled from the [Recording Settings](#) page, voice announcements will be played in the conference to inform participants that the conference recording is started, stopped, paused or resumed.

Recorded conferences are stored and are listed in the **Recorded Conferences** page accessible by the moderator from QuadroCS Web Management.

The **Recorded Conferences** page displays a table where recorded conferences are listed. The recorded conferences can be played, deleted and saved from this page.

**Recording free space** shows the free space allocated for the corresponding conference.

The **Recorded Conferences** table displays all the recorded conferences with the following parameters:

**Date & Time** shows the initiation date and time of the recorded conference.

**Duration** shows the duration of the recorded conference (in minutes/seconds).

**Play** - by clicking on the speaker sign beside every record in the table, the recorded conference will be played (using the available media player supported by your Operation System).

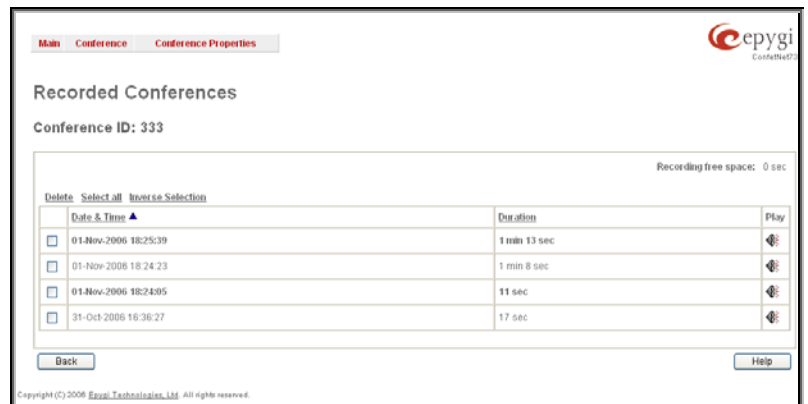


Fig. III-7: Recorded Conference

The column headings of the **Recorded Conferences** table are organized as links. By clicking on the column heading, the table will be sorted by the selected column. Upon sorting (ascending or descending), arrows will appear next to the column heading. Each row in the table of Recorded Conferences can be selected by the checkbox for deletion.

**To Play a Conference**

1. Click on the speaker sign of the corresponding recorded conference.
2. Depending on your browser settings, the .wav file will be played directly or an application will ask you to save the .wav file locally to the PC. If you need to save the file, please specify the path then run the media file from the specified location.

**To Delete a Recorded Conference**

1. Select the checkbox of the corresponding record(s) in the **Recorded Conferences** table that will be deleted. Click on **Select all** if all records should be deleted.
2. Select the **Delete** button.
3. Confirm the deletion clicking **Yes**. The selected conference then will be deleted. To abort the deletion and keep the conference on the QuadroCS, select **No**.

**Conference Statistics**

The **Statistics** page consists of three pages with conference statistics on the first page, successful call statistics on the second page and nonsuccessful call statistics on the third page. The Statistics allows collecting conference call events on the QuadroCS with their parameters and to search them by various criteria.

The **Conferences** page displays a table with detailed information (Activation Time, Conference Duration, Participants Count, Activation Reason and Activation Details) about the established conferences and gives the possibility to view the component calls of a particular conference.

To do so, click on the corresponding conference ID. To view all calls directed to or dialed out of the QuadroCS, move to the **Successful Calls** page where the call details table (Activation Time, Conference Duration, Participant, Call Start Time and Call Duration) is displayed. Failed calls can be viewed in the **Nonsuccessful Calls** page.

The **Conferences** page displays a table with detailed information (**Activation Time, Conference Duration, Participants Count, Activation Reason and Activation Details**) about the established conferences and allows viewing the individual conference calls.

To use this feature, click on the corresponding conference ID. To view all calls directed into or dialed out of the QuadroCS, move to the **Successful Calls** page where a table shows the call details (**Activation Time, Conference Duration, Participant, Call Start Time and Call Duration**).

Each column heading in the table is a link. By clicking on the column heading, the table will be sorted by the selected column. Upon sorting (ascending or descending), arrows will be displayed beside the column heading.

For both **Conferences** and **Successful Calls** pages, a statistics summary and an option to search components are available.

The statistics summary contains the **Number or (Call) Records**, which displays the total number of conference entries or successful calls in the corresponding tables. Additionally, **Conf Total Duration, Conf Maximum Duration, Conf Average Duration and Conf Minimum Duration** columns provide information about the total, maximum, average and minimum durations of the conferences.

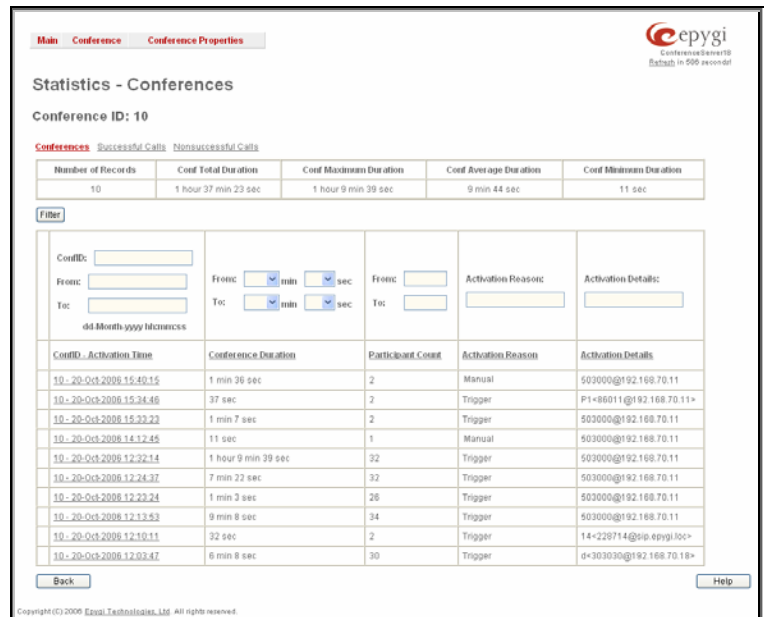


Fig. II-8: Call Statistics Page

**Search components** allow the user to filter the **Conferences** or **Successful Calls** tables:

The **ConfID** text field requires the ID of the conference call. Only numeric values are allowed in this field.

The **From** and **To** text fields above the **Activation Time** column are used to search the conferences by the conference activation date and time. The data must be entered in the following format: dd-mm-yyyy hh:mm:ss. The time criterion is optional. The **From** field requires an earlier date and time than the **To** field.

The **From** and **To** text fields above the **Participant Count** column are used to search by the number of participants in the conference. You can perform manual, trigger or scheduled searches of conferences by using the **Activation Reason** text field. Conferences can be searched by their activation details by using the **Activation Details** text field. The value in the **Activation Details** column is dependent on the **Activation Reason**. For **Manual** activation, details can be either **Admin** or **Moderator** indicating the initiator of the conference. For **Trigger** activation, details indicate the callers' address. For **Schedule** activation, details indicate the conference activation date and time.

By using the **Participant** text field located in the **Successful Calls** table call participants can be searched by their SIP addresses (see chapter [Entering a SIP Addresses correctly](#)). If the defined SIP address is inserted incorrectly, the error "Participant address is incorrect" will prevent filtering. The **From** and **To** text fields above the **Call Start Time** column in the **Successful Calls** table are used to search incoming or outgoing calls by the call start date and time. The data has to be entered in the following format: dd-mm-yyyy hh:mm:ss. The time criteria are optional. **From** requires an earlier date and time than the **To** field. If the entered data does not meet this condition, the error message "Minimal date should be less than maximal date" prevents statistics filtering.

The **From** and **To** drop down lists above the **Call Duration** column in the **Successful Calls** table are used to search by the call duration. The duration has to be selected from the listed values. The **From** field has to indicate a shorter duration than the **To** field. If the inserted data does not meet this condition, the error message "Minimal duration should be less than maximal duration" prevents statistics filtering.

In the **Nonsuccessful Calls** page, entries can be searched by the Conference ID, by the participant's SIP address, the call start time and the call details (this column indicates a reason of the call failure).

**Filter** performs a search procedure by the selected criteria. The search may be done with several criteria at once.

**To Filter the Statistics**

1. Enter the desired criteria fields.
2. Press the **Filter** button to search the call reports within the **Statistics** table.

**Please Note:** To return to the complete **Statistics Table** clear all search criteria and press **Filter**.

## Conference Properties Menu



Fig. III-9: Moderator Settings Dynamo Menu



Fig. III-10: Moderator's Settings Tree Menu

## General Settings

The **Conference General Settings** page is used to configure the basic conference settings.

**Please Note:** This page is read-only and the **Password** fields are hidden for the users authenticated as participants.

The **Conference General Settings** page contains the following components:

**Subject** is an optional parameter where a brief subject of the conference can be entered.

**Description** is an optional parameter used to insert additional information about the conference.

**Moderator Password** can be entered to provide moderator access to the conference. It has to be entered twice for confirmation. The password inserted here will be used by the moderator to join the conference. The moderator can dial codes during the active conference as well as access conference specific GUI pages to control the conference (view/change conference properties, activate/deactivate it, start/stop/resume record, view conference statistics).

**Participant Password** can be entered to require a password for participant access to the conference. It has to be entered twice for confirmation. The password entered here should be used by the participant to join the conference. The participant can participate in the conference only according to the rights (speaker or listener) granted by the moderator. The participant also can access the conference specific GUI to view the conference properties and the other participants.

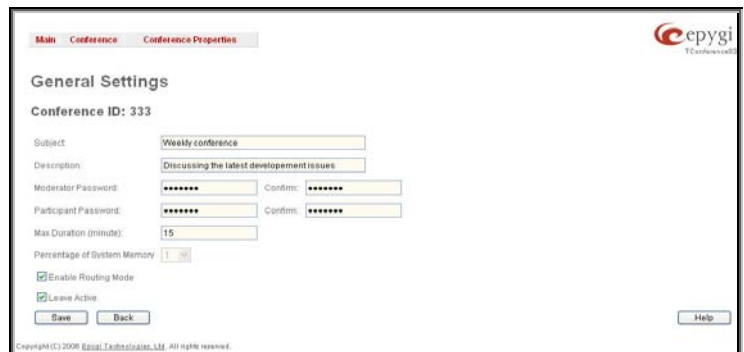


Fig. II-11: General Settings Page

**Max. Duration** sets the conference to be limited to a maximum duration (in minutes). Leave the field empty for unlimited conference duration.

**Percentage of System Memory** is used to select the percentage of total QuadroCS's memory space that can be used for recording of this conference.

**Enable Routing Mode** checkbox enables direct access to the Quadro's internal routing tables when dialing out to a participant during a conference. When this checkbox is selected, any dialed number will be parsed through Call Routing Table (see Manual-II: Administrator's Guide).

**Leave Active** checkbox will keep conference active, even if all participants have left it.

## SIP Registration Settings

The **SIP Registration Settings** page is used to manage the SIP registration settings of the conference. SIP Registration Server and SIP Outbound Proxy Server settings may be configured here. The SIP server registration is needed for conference to be reachable in the IP network.

**Please Note:** This page is read-only for the users authenticated as participants.

This page offers the following components:

**User Name** text field requires a user name for conference registration on the SIP server. The registration user name needs to be unique on the SIP server and is used to reach the conference over the SIP network.

**Password** indicates the password for the extension registration on a SIP server.

**Confirm Password** is for password confirmation. If the entered password does not correspond to the one given in the **Password** field, an error will appear.

**SIP Server** indicates the host address of the SIP server. The field is not limited regarding symbol usage and length as it can be either an IP address, for example, 192.168.0.26 or a host address, e.g., sip.epygi.com.

**SIP Port** indicates the host port number to connect to the SIP server. The SIP server port may contain only numeric values. If the SIP server port is not specified, QuadroCS will access the SIP server through the default port 5060.

**Registration on SIP Server** enables conference registration on the SIP server registration.

The link [SIP Advanced Settings](#) moves to a page where advanced SIP settings (**Outbound Proxy**, **Secondary SIP Server** and **Outbound Proxy for the Secondary SIP Server** settings and other SIP server specific settings) can be configured.

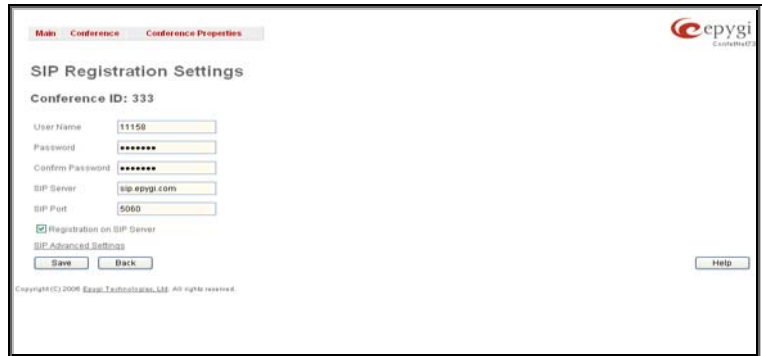


Fig. II-12: SIP Registration Settings Page

## SIP Advanced Settings

The SIP Advanced Settings page is used to configure advanced SIP settings (Outbound Proxy, Secondary SIP Server and Outbound Proxy for the Secondary SIP Server settings and to define other SIP server specific settings).

The SIP Outbound proxy is an SIP server where all the SIP requests and other SIP messages are transferred. Some SIP servers use an outbound proxy server to escape restrictions of NAT. For example, Free World Dialup service uses an Outbound Proxy server. If an Outbound proxy is specified for a conference, all dial-out SIP calls originating from that conference are made through that outbound proxy, i.e., all requests are sent to that outbound proxy, even those made by Speed Calling.

The Secondary SIP Server acts as an alternative SIP registration server when the primary SIP Registration Server is inaccessible. If the connection with the primary SIP server fails, Quadro will automatically start sending SIP messages to the Secondary SIP Server. It will switch back to the primary SIP server as soon as the connection is reestablished.

**Authentication User Name** requires an identification parameter to reach the SIP server. It should be provided by the SIP service provider and can be requested for some SIP servers only. For others, the field should be left empty.

**Send Keep-alive Messages to Proxy** enables the SIP registration server accessibility to the verification mechanism. **Timeout** indicates the timeout between two attempts for the SIP registration server accessibility verification. If no reply is received from the primary SIP server within this timeout, the Secondary SIP server will be contacted. When the primary SIP server recovers, SIP packets will resume being sent to it.

The **RTP Priority Level** drop down list is used to select the priority (low, medium or high) of the RTP packets sent from a corresponding conference. RTP packets with higher priority will be sent first in case of heavy traffic.

A group of **Host address** and **Port** text fields respectively require the host address (IP address or the host name) and the port numbers of the **Outbound Proxy**, **Secondary SIP Server** and the **Outbound Proxy for the Secondary SIP Server**. These settings are provided by the SIP servers' providers and are used by Quadro to reach the selected SIP servers.

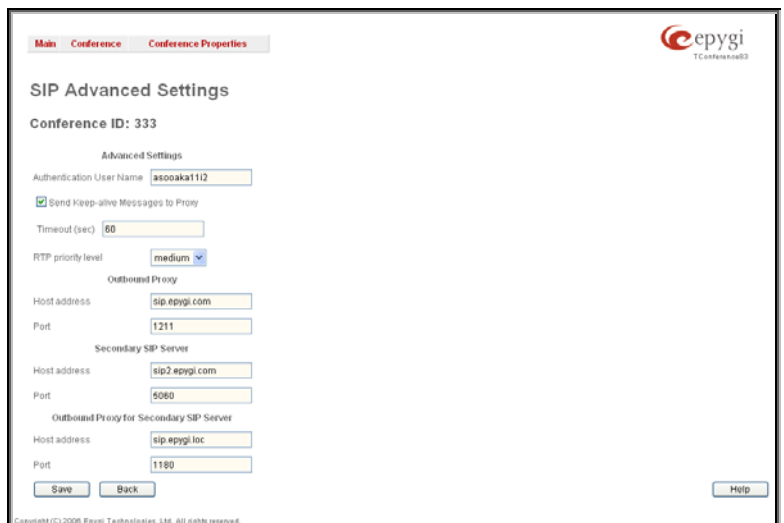


Fig. II-13: SIP Advanced Settings page

## Recording Settings

The settings on this page are addressed to the conference recording configuration, enabling conference recording, defining the recording memory allocation (internal storage or the external USB flash), selecting a recording indication type, etc.

**Please Note:** This page is read-only for the users authenticated as participants.

The **Conference Recording Settings** page offers the following components:

**Enable Recording** checkbox enables an option to be used for active conferences to perform the online recordings. With this checkbox selected, a group of radio buttons is activated to select the storage for the recorded conference audio files.

**Use Internal Storage** switches the location that is used to store the recorded conference audio files to the QuadroCS's internal memory. Following settings are required to be adjusted for this selection:

- **Recording Codec** drop down list requires selecting the audio codec that will be used for conference recordings. The recording codec selection will affect recorded voice quality and the output file size.
- **Max Recording Time** requires the maximum duration (in minutes) of one recording to be done. If the conference recording has been paused and resumed again, the Max Recording Time value will indicate the actual recorded time. Leave this field empty not to limit the duration of the conference recording.

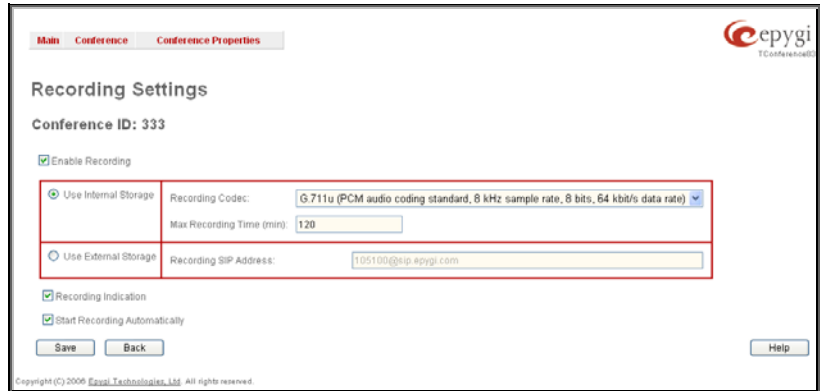


Fig. II-14: Recording Settings page

**Use External Storage** switches the location used to store the recorded conference audio files to an external destination, which can be any device or application that has audio recording capabilities. The **SIP Address** of the remote destination where the recorded conference will be stored is required to be defined for this selection. Optionally, the SIP address of a user can be inserted here. In this case, the conference will be recorded to the private mailbox of the user or will be directly played to him if he answers the incoming call from the QuadroCS.

**Recording Indication** selection enables voice announcements played in the conference to inform participants that the conference recording is started, stopped, paused or resumed.

When the **Start Recording Automatically** checkbox is selected, the conference recording will automatically start as soon as the corresponding conference is activated.

## Customization

The **Customization** page is used to manage the voice prompts played during the active conference. The page offers the following options:

When the **Play First in Conference message** checkbox is selected, the QuadroCS will play a "You are the first participant in the conference" notification message informing you that no more participants are yet connected.

**Welcome Message** parameters group allows updating the active conference welcome message (played once a user is connected to the conference), downloading it to the PC or removing the custom welcome message. The group offers the following components:

**Upload new welcome message** indicates the file name used to upload a new welcome message. The uploaded file needs to be in PCMU wave format, otherwise the system will prevent uploading it and the "Invalid audio file, or format is not supported" warning message will appear. The system also prevents uploading if there is not enough memory available for the corresponding conference and the "You do not have enough space" warning message will appear.

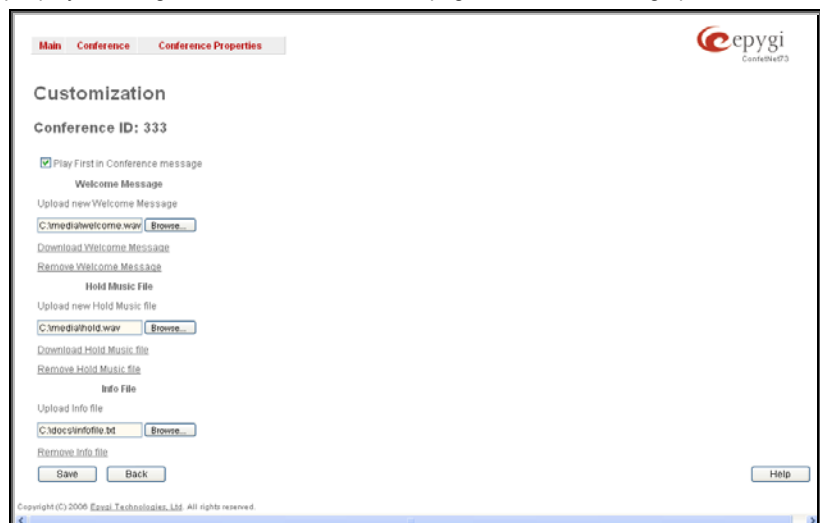


Fig. II-15: System Messages page

**Browse** opens the file chooser window to browse for a new welcome message file.

The **Download Welcome Message** and **Remove Welcome Message** links appear only if a file has been uploaded previously. The **Download Welcome Message** link is used to download the message file to the PC and opens the file-chooser window where the saving location may be specified. The **Remove Welcome Message** link is used to restore the default welcome message.



**Hold Music File** parameters group allows updating the hold music (played when you are alone in the conference), downloading it to the PC or removing the custom welcome message. The group offers the following components:

**Upload new hold music file** indicates the file name used to upload a new hold music file. The uploaded file needs to be in PCMU wave format, otherwise the system will prevent uploading it and the "Invalid audio file, or format is not supported" warning message will appear. The system also prevents uploading if there is not enough memory available for the corresponding conference and the "You do not have enough space" warning message will appear.

**Browse** opens the file chooser window to browse for a new hold music file.

The **Download Hold Music File** and **Remove Hold Music File** links appear only if a file has been uploaded previously. The **Download Hold Music File** link is used to download the hold music file to the PC and opens the file-chooser window where the saving location may be specified. The **Remove Hold Music File** link is used to restore the default hold music.

**Info File** parameters group allows you to upload a text file with some conference related announcement, advertisement or any other information to be displayed on the [Conference Progress](#) page. The group offers the following components:

**Upload Info file** indicates the information file name. The system will display the file content exactly in the way it is formatted in the file. It is recommended to use a \*.txt formatted plain text file. The uploaded file should not exceed the size of 2000 bytes. The system also prevents uploading if there is not enough memory available for the corresponding conference and the "You do not have enough space" warning message will appear.

**Browse** opens the file chooser window to browse for an information file.

The **Remove Info File** link appears only when a file has been previously uploaded and is used to remove the uploaded information file.

## Participants

This page allows to configure participants of the conference as well as to adjust settings of the participants dialed out during the conference or independently connected to the conference.

The [New Participants Configuration](#) moves to the page where the settings of participants independently dialed in to the conference can be configured. Once the new participant connects the conference, he will automatically appear in the [Conference Progress](#) table on this page and remain there unless disconnected from the conference.

The Handset Added Participants Configuration moves to the page where the settings of participants dialed out from the handset by the moderator during the active conference can be configured. Once the handset added participant connects the conference, he will automatically be added to the [Conference Progress](#) table on this page and remain there unless the conference is terminated.

The table on this page lists all preconfigured participants, allows to add new participants and to modify the settings of the exiting ones.

**Please Note:** The table on this page is displayed in a read-only mode for the users authenticated as participants.

**Participants**  
**Conference ID: 10**

[New Participants Configuration](#)  
[Handset Added Participants Configuration](#)

Registered Participants:  
[Add](#) [Edit](#) [Delete](#) [Select all](#) [Inverse Selection](#)

	Name	SIP Address	Participant Type	Dial Out	Trigger	Participant Indication	Confirmation Type	Nested Conference	Duplication	Mail Address
<input type="checkbox"/>		228714@sip.epygi.loc	Speaker	Yes	Yes	Yes	None	No	No	
<input type="checkbox"/>		224813@sip.epygi.loc	Speaker	Yes	Yes	Yes	None	Yes	No	
<input type="checkbox"/>	Diana	20233@sip.epygi.com	Listener Only	No	Yes	Yes	Password	Yes	No	diana@epygi.com

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Fig. II-16: Participants Page

**Add** opens an **Add Entry** page where new participants can be added to the conference. The following parameters are needed to configure participant settings:

**Participant Name** requires optional information (first name, last name, nickname, etc.) about the participant.

**SIP Address/Tel. number** requires the contact phone number (SIP address or Routing Number) of the participant. This number automatically will be dialed by the QuadroCS when the participant is configured to be a Dial Out (see below) or when a corresponding Conference Code is used (see Conference Codes).

The participant's SIP address should be a combination of username@hostaddress:port (where hostaddress can be an IP address, for example, 192.168.90.10, or a host name, e.g., sip.epygi.com). The port number is optional for the SIP address. If no port is specified, 5060 will be used. The range of valid ports is between 1024 and 65536.

The value will be implied as a Routing Number when either digits starting with zero (0) are inserted in this field or Routing Mode is enabled for the conference. In these cases, the Routing Number will be parsed through the Local Routing Table when placing a call to the corresponding participant. Otherwise, when Routing Mode is disabled, any numerical value starting with a non-zero digit will be treated as a SIP username and will be applied with the host name of the SIP server the conference is currently registered on.

**Please Note:** A direct call will be placed toward a participant's SIP address if the corresponding conference is registered on a different SIP server than the participant is registered on, or if the participant is not registered on the SIP server at all.

**Email Address** requires the email address of the participant. Conference related notifications (configured from the [Schedule](#) page or using the [Send Notification Mail](#) option) will be sent automatically to this address.

**Participant Type** list is used to select the type (speaker or listener) of participant in the corresponding conference.

**Confirmation Type** list is used to set the password protection for the participant joining the active conference. **Star (\*)** selection allows the participant to accept the conference invitation by pressing the \* button. Only participants connected to the conference with the moderator password will be provided with permissions to manipulate the conference.

A group of checkboxes on this page allow configuration of participant specific settings related to the corresponding conference:

- When the **Dial Out** checkbox is selected, the certain participant will be automatically dialed out when the conference is activated.
- **Activate On Dial In** automatically activates the corresponding conference when this participant joins the conference call.
- **Participant Indication** enables the beep indication during the conference when this participant joins or leaves the conference.
- **Nested Conference** must be selected if the participant is a Conference itself and enables the correct behavior of conference termination.
- **Allow Duplicated Participation** checkbox allows multiple participants with the selected Caller ID (calling address) to join the corresponding conference. This is applicable when different participants are using the same shared number to place a call.

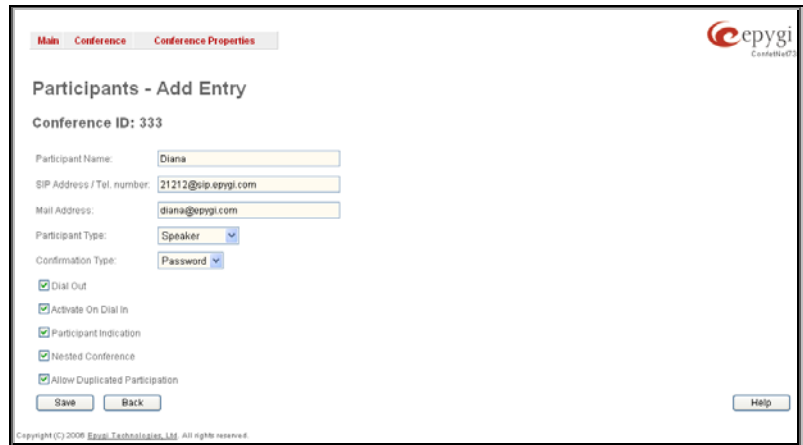


Fig. II-17: Participants - Add Entry Page

The **Edit** functional button provides a possibility of editing multiple participants at the same time. A **Select to modify fields** checkbox alongside the fields to be modified needs to be selected to submit changes, otherwise the fields will not be updated.

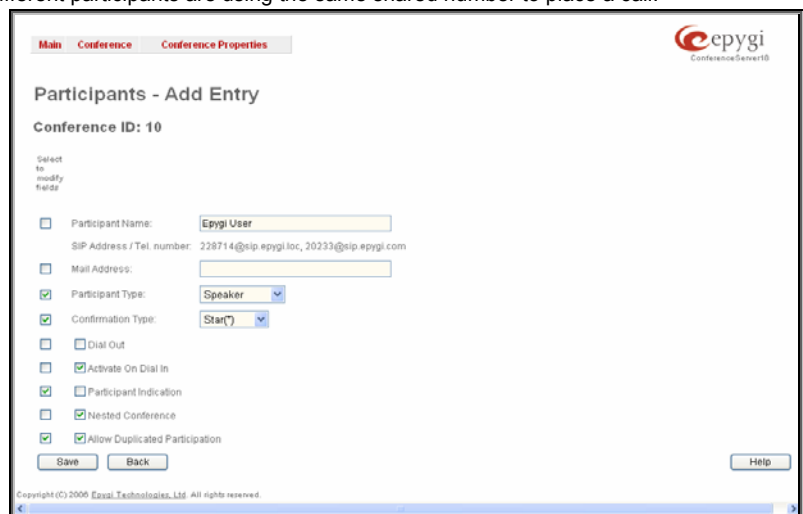


Fig. II-18: Participants – Multi-Edit Entry Page



## New Participants Configuration

This page is used to configure settings of participants independently dialed in to the conference. Once the new participant connects the conference, he will automatically appear in the [Conference Progress](#) table and remain there unless disconnected from the conference.

**Please Note:** The settings on this page are displayed in a read-only mode for the users authenticated as participants.

The page consists of the following components:

Selecting the **New Participant Allowed to Join** checkbox will allow new users to connect the conference by simply dialing in (no previous registration in [Participants](#) table is needed). If this checkbox is selected, the following settings should be provided:

**Max New Participant Count** text field requires the maximum number of new users allowed to connect to the conference. Leave this field empty to allow unlimited number of new users connecting the conference.

**New Participant Type** drop down list is used to select the state (speaker or listener only) of the new participants connected to the conference.

**New Participant Confirmation Type** drop down list is used to select whether the conference is password protected for the new users or not.

Selecting the **New Participant Can Activate Conference** checkbox will allow the new users to activate the conference.

Selecting the **New Participant Indication** checkbox will enable a beep indication during the active conference when a new user joins or leaves the conference.

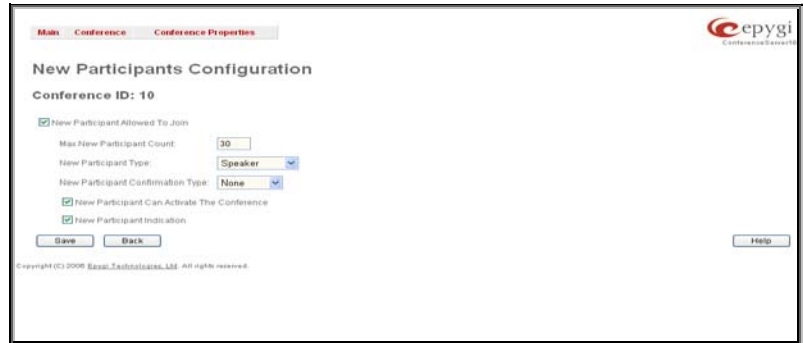


Fig. II-19: New Participants Configuration Page

## Handset Added Participants Configuration

This page is used to configure the settings of participants dialed out from the handset by the moderator during the active conference. Once the handset added participant connects the conference, he will automatically appear in the [Conference Progress](#) table and remain there unless the conference is terminated. This will allow the handset dialed participant to hang up and dial in to the corresponding conference again while it is active.

**Please Note:** The settings on this page are displayed in a read-only mode for the users authenticated as participants.

The page consists of the following components:

**Participant Type** drop down list is used to select the state (speaker or listener only) of the handset added participants connected to the conference.

**Confirmation Type** drop down list is used to select whether the conference is password protected for the handset added users or not. When **Star (\*)** selection is chosen, the handset added user should accept the conference invitation by pressing the \* button.

Selecting the **Participant Indication** checkbox will enable a beep indication during the active conference when a handset added user joins or leaves the conference.

The **Allow Duplicated Participation** checkbox selection allows several instances of callers with the same handset added number (caller address) to join the corresponding conference at the same time. This option may be used to allow users from the same network (with the same caller address), like E1/T1 network, to reach the conference.

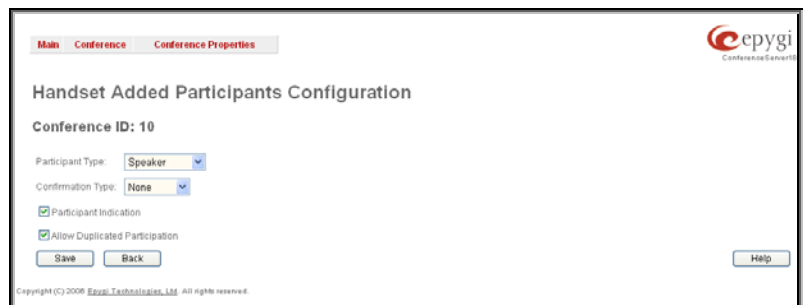


Fig. II-20: New Participants Configuration Page

## Schedule

The **Schedule** page is used to configure and manage the conference scheduling rules, so that a conference can be automatically activated on the date and time. The Scheduling service may also be configured to send invitation emails to the participants asking them to join the conference or informing about a new conference.

**Please Note:** The table on this page is displayed in a read-only mode for the users authenticated as participants.

The **Conference Schedule** page offers a table that lists all scheduling rules configured for the corresponding conference. When a scheduled conference is activated, all participants with dial-out option enabled will be dialed.

Clicking the **Add** button takes you to the **Add Entry** page where new scheduling rule can be configured. This page offers the following components:

A group of radio buttons that are used for selecting the frequency of the scheduled conference:

- **Once** – the calendar date (month, day, year) should be specified for this option.
- **Daily**
- **Weekly** – weekdays when scheduling out to be activates should be selected for this option. Use **Select All** and **Select None** to select or deselect all weekdays.
- **Monthly** – the calendar day should be selected for this option.
- **Annually** – the calendar day and the month should be selected for this option.

In the **Time** text fields, the time of the scheduled conference activation should be defined. The time selected in these fields will be considered according to the global Time/Date Settings configured on the QuadroCS (see Manual II – Administrator's Guide).

The **Allow Participants to join conference before Conference Activation** checkbox selection allows participants to dial in to the conference before conference activation. During this period, participants will be able to communicate with each other. However, this does not mean that the conference is activated; the participants will be dialed out (if any) and the recording will start (if configured) only after the configured scheduled time comes.

The **Send Mail before Conference Activation** checkbox enables email notification delivery to the participants before the conference activation. The text field requires the timeout (in minutes) before the conference activation when the email notifications to the conference participants with **Email Address** configured from the **Add Participants** page should be delivered. This option is only valid if the Email Address is configured for the participant.

The **Send Mail on behalf of** text field requires an email address or a conditional name related to the conference to be transmitted in the **From** field of the email notifications.

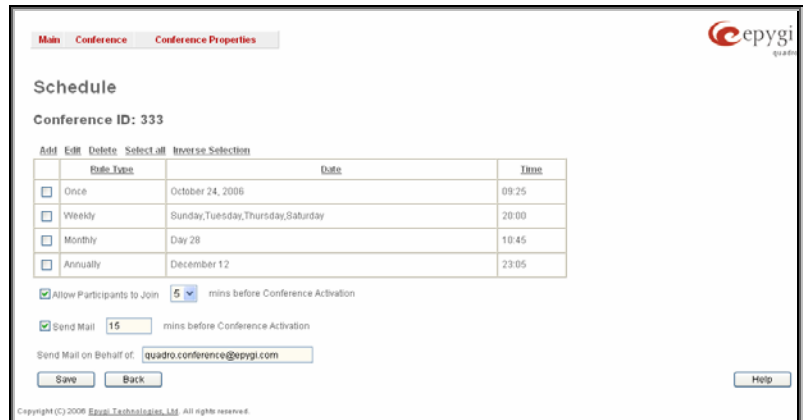


Fig. II-21: Schedule Page

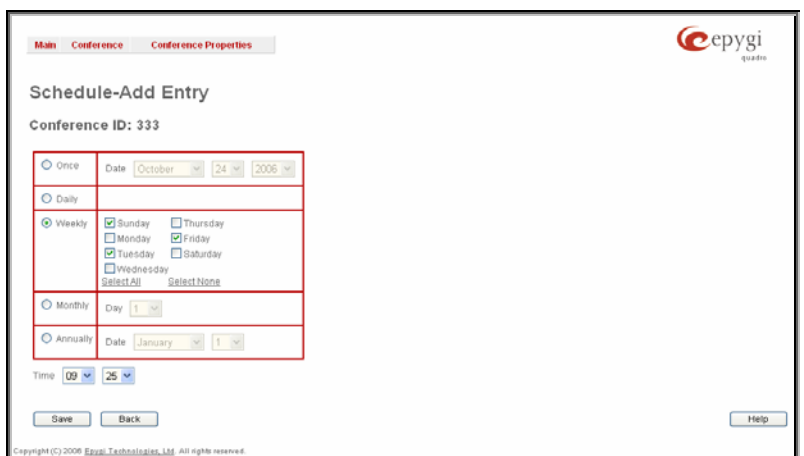


Fig. II-22: Schedule – Add Entry page

## Reservation

The **Reservation** service allows you to reserve the conference resources having a limitation of 30 participants connected to the QuadroCS at the same time. This service is used to guarantee the number of participants in a certain conference, otherwise not all important participants might be able to connect the conference if there are other conferences running.

When a certain number of participants are reserved for the fixed time frame, any other conferences will not start or additional participants will not be able to connect the conferences if there are no enough resources on the QuadroCS. Following rules are applicable to the reservation mechanism:

- Reserved conferences have a higher priority versus to the non-reserved conferences. In case if no enough resources are available on the QuadroCS, only non-reserved conference will be terminated.
- If a Conference1 is running on the QuadroCS with more participant (e.g. 7) than the upcoming reservation of the same conference allows (e.g. 5), the previously started Conference1 will be automatically terminated when a reservation time comes. Further, a new conference may start with participants no more than allowed by the reservation (e.g. 5).
- If a non-reserved Conference1 is running on the QuadroCS with the certain number of participants (e.g. 10) and a time comes for reservation of the Conference2 with the number of participants (e.g. 5), the Conference1 will keep running if there are enough resources on the QuadroCS.
- If a Conference1 running on the QuadroCS with the certain number of participants (e.g. 28, in particular case this number may be reached by a number of multiple conferences active at the same time) and a time comes for the reservation of the Conference2 with the number of participants (e.g. 5), the Conference1 will be automatically terminated because the total resources (e.g. 28+5) will exceed the limitation of 30 participants (or in case if a number of multiple conferences are active, only not reserved conferences will be terminated while the reserved ones will keep running).
- If there is already a conference running with fewer participants than it is allowed by the reservation entry, the previously started conference will keep running even after the reservation time comes and additional participants will be able to connect the conference, as far as the number of participants configured by the reservation is not reached.

The **Reservation** page is used to reserve the conference resources for the single time period. From this page a chart with conference resources usage can be generated.

**Please Note:** The table on this page is displayed in a read-only mode for the users authenticated as participants.

The table on this page lists all reservations configured for the corresponding conference. When the reservation period passes, the reservation automatically disappears from the **Reservations** table.

Clicking the **Add** button takes you to the **Add Reservation** page where a new reservation can be made for a conference. This page offers the following components:

The **Start Date** drop-down lists are used to select the month, the day and the year when the conference reservation should start.

The **Start Time** drop-down lists are used to select the time when the conference reservation should start.

The **Duration** text field requires the duration (in minutes) of the reserved conference.

The **Participant Count** drop-down list is used to select the maximum number of participants in the reserved conference.

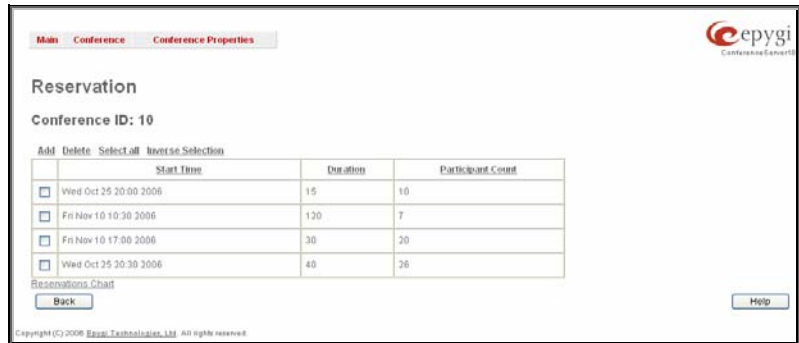


Fig. II-23: Schedule Page



Fig. II-24: Reservation – Add Entry page



Fig. II-25: Reservations Chart configuration page

The **Reservations Chart** link takes you to the page where chart characteristics (the start date, the start time and the duration) should be defined. Once the characteristics are defined, press **Show** to display the reservation chart.

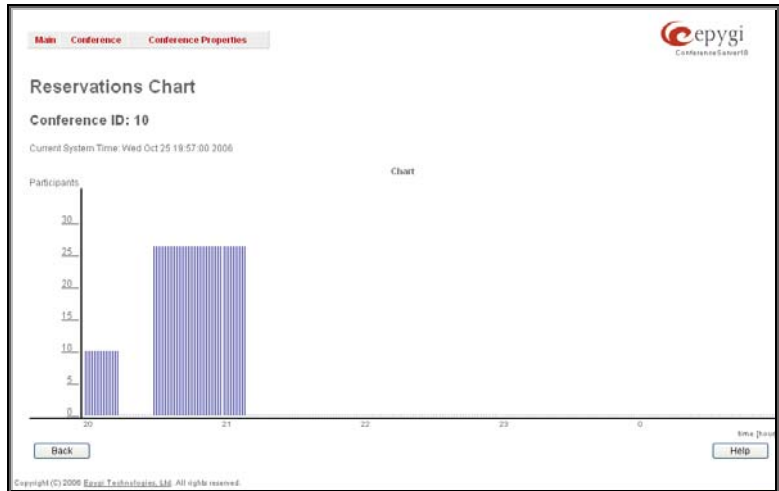


Fig. II-26: Reservation Chart

The **Reservations Chart** displayed on this page illustrates the dependency of the reserved number of participants (vertical axis) on the time variable (horizontal axis).

### Codecs

To establish IP voice communication, both parties have to use the same codec. During setup of the communication line, the codec is negotiated. If the caller does not find a matching codec, the communication won't start. If you want to be reachable by all IP callers, it is helpful to support as many codecs as possible. In this case, all codecs that Quadro supports should be added to the **Active Codecs**' table. On the other hand, some codecs require significant bandwidth- up to 64 kBit/s. If you definitely don't want to use these codecs, you must remove them from the **Active Codecs**' table.

The **Conference Codecs** page displays a list of Quadro **Active Codecs** used to initiate the conference.

The table **Active Codecs** lists active voice codecs for the selected line that are supported by Quadro. The order of records in the **Active Codecs**' table is important for transmitting and receiving. A codec placed at the top of the table will be used as the preferred codec. If the remote party does not support the preferred codec, the following codecs will be tried out strictly in the top down order given in the **Active Codecs**' table.

Each record in the table has an assigned checkbox. It is used to select the record to be deleted or moved up or down. The following error message results if no records are selected and the user activates the delete button: "No records selected". At least one codec must be attached to the conference, when attempting to delete the last one the error will occur: "At least one codec should stay in the codec list".

**Please Note:** The table on this page is a read-only mode for the users authenticated as participants.

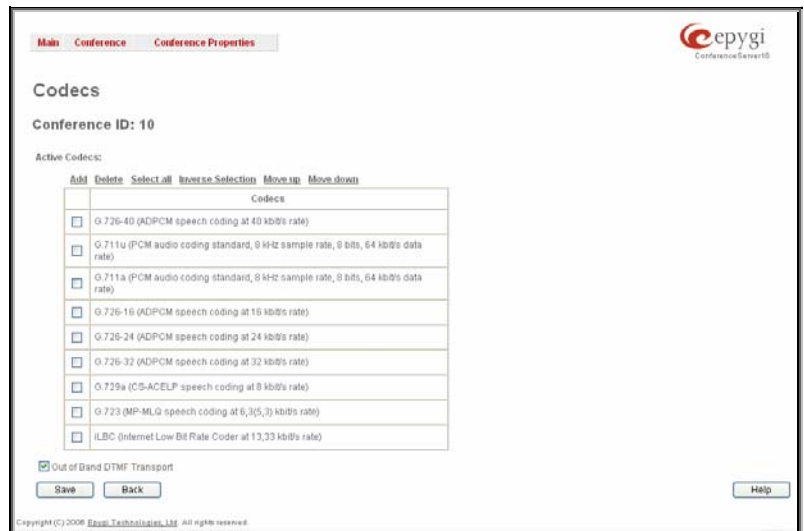


Fig. II-27 Codecs page

**Add** opens the **Add Entry** page where the user may add codecs supported by QuadroCS. The voice codec defines the voice compression algorithm for the incoming and outgoing DSP packages.

**Codecs** lists all codecs supported by Quadro. If no more codecs are available (all available codecs have already been transferred to the **Active Codecs**' table), the **Add Entry** page will display the message "No Available Codecs" instead of the drop down menu.



Fig. II-28 Codecs – Add Codec page

The **Move Up/Move Down** buttons are used to move the selected codec one level up or down in the table.

The **Out of Band DTMF Transport** checkbox enables DTMF code transmission in parallel with the voice stream. The destination receiving the DTMF code will play it locally if it supports the feature. This is helpful to avoid DTMF's loss upon bad traffic. This feature is valuable for all codecs but it is especially recommended to enable it in case low bit rate codecs (G729, G723, G726/16, etc.) are selected.

## AGC Settings

**Admission Gain Control** option is used to equalize the voice levels of all participants in the conference. In case if some participants have too low voice level while others sound too loud, this option is used to adjust all voices to some average level before sending them to the conference.

The **AGC Settings** page is used to enable and configure the **Admission Gain Control** parameters.

**Enable AGC** checkbox is used to enable the **Admission Gain Control** option on the corresponding conference.

**Low Threshold** and **High Threshold** fields indicate the range of the acceptable voice level in the conference to which any low or high voices out of this range should be adjusted. Fields may have values in a range between -50 and 0 dB.



Fig. II-29 AGC Settings page

## Logout

This option is used to close the session between the user PC and QuadroCS and to leave web management or to re-enter web management with another login. By selecting the **Logout** button, the startup page will be displayed and the user will need to login again.

## Feature Codes

This chapter describes how the Quadro Conference Server codes allow the moderator and participants to modify the conference environment with the handset.

### Conference Services accessible during the conference:

<p><b>Invite Participant</b></p> <p>To invite a participant dial * 1 + Participant's SIP address (or * 1 + Routing Number if the Routing Mode is enabled). This service is available for moderators only.</p>	* 1
<p><b>Get the number of participants in the conference</b></p> <p>Plays information about the total number of participants in the conference at the certain moment.</p>	* 2 1
<p><b>Get the state of recording</b></p> <p>Plays the state of conference recording (started, stopped or paused).</p>	* 2 2
<p><b>Lock the conference</b></p> <p>Locks the conference. When the conference is locked, nobody can dial in during lockout. This service is available for moderators only.</p>	* 3 1
<p><b>Unlock the conference</b></p> <p>Unlocks the conference. Now participants are allowed to dial in to the conference. This service is available for moderators only.</p>	* 3 2
<p><b>Dial out to all users with dial out settings enabled</b></p> <p>Initiates the dial-out to all participants currently inactive in the conference but configured to be dialed out (also those added manually from the handset by moderator). This service is available for moderators only.</p>	* 4 1
<p><b>Dial out to all users participant to the conference</b></p> <p>Initiates the dial-out to all participants currently inactive in the conference. This service is available for moderators only.</p>	* 4 2
<p><b>Start or Resume Conference Recording</b></p> <p>This service is available for moderators only.</p>	* 8 1
<p><b>Pause Conference Recording</b></p> <p>This service is available for moderators only.</p>	* 8 2
<p><b>Stop Conference Recording</b></p> <p>This service is available for moderators only.</p>	* 8 3
<p><b>Request to Speak</b></p> <p>With this key combination, listener requests to speak and a notification hand-up icon is displayed in the Conference Progress table. The moderator can then switch the particular listener either to speaker or lecture mode. With a speaker permission granted, listener can speak to the conference along with other participants. With a lecturer permission granted, listener can speak to the conference having all other participants muted in the conference. This service is available for listener participants only.</p>	* 9 1
<p><b>Cancel the Request to Speak</b></p> <p>With this key combination, listener cancels his request to speak and a notification hand-up icon disappears from the Conference Progress table. This service is available for listener participants only.</p>	* 9 2

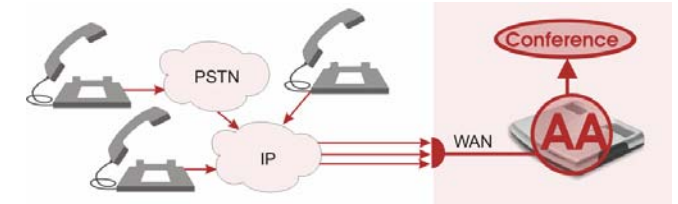
**Please Note:** You may accelerate dial out by a pound (#) sign at the end of your dialed number.

## QuadroCS' Auto Attendant Services

**Conference Server's Auto Attendant** provides remote access to the QuadroCS conferences. Specifically it allows to join the conference and to use the Zero Out service (if configured). **Conference Server's Auto Attendant** can be accessed remotely from the IP network by dialing Auto Attendant's SIP address. The Auto Attendant has its own entry in the Conference Management table. In order to receive IP calls, the entry dedicated to Auto Attendant needs to be registered at a SIP server.

The automated attendant services are divided into two feature groups: **Conferences** menu and **Zero Out** menu. Voice messages help the caller to navigate within those areas using handset buttons.

The **Conferences Menu** is used to locally access the conferences on the Quadro Conference Server. Depending on conference configuration, password may be asked to access the conference.



The **Zero Out Service** provides call redirection to the configured destination (can be reception, operator, management, etc.).



### Call Codes available in the Auto Attendant:

For external IP calls addressed to the Auto Attendant following key combinations are available to access and manipulate within Auto Attendant services:

Entering the Auto Attendant Services	Key Combination
<b>Conferences Menu</b> - used to access conferences. <b>Conference ID</b> should be dialed here.	already in
<b>Zero Out</b> - provides automatic call redirection (if enabled from Auto Attendant Settings page)	0



## Appendix: Glossary

### A

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**Asymmetric Digital Subscriber Line (ADSL)** - is a method for moving data over regular phone lines. An ADSL circuit is much faster than a regular phone connection, and the wires coming into the subscriber's premises are the same (copper) wires used for regular phone service. An ADSL circuit must be configured to connect two specific locations, similar to a leased line. A commonly discussed configuration of ADSL would allow a subscriber to receive data (download) at speeds of up to 1.544 Megabits per second, and to send (upload) data at speeds of 128 kilobits per second. Thus the 'Asymmetric' part of the acronym. Another commonly discussed configuration would be symmetrical: 384 kilobits per second in both directions. In theory ADSL allows download speeds of up to 9 megabits per second and upload speeds of up to 640 kilobits per second. ADSL is often discussed as an alternative to ISDN, allowing higher speeds in cases where the connection is always to the same place.

**Asynchronous Transfer Mode (ATM)** - a 53-byte cell-switching technology well suited for carrying voice, data, and video traffic on the same infrastructure. It is inherently scalable in throughput and was designed to provide Quality of Service (QoS).

**Auto Attendant (AA)** - a feature providing remote access to Quadro voice connectivity services. Specifically, it supports remote connection to Quadro extensions, to their mailboxes and for making calls to other destinations. Remote access to Quadro AA is possible through IP and PSTN calls.

**Auto Redial** - a service that allows automatically recalling the destination that was busy.

### C

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**Call** - establishment of (or attempt to establish) a voice or data connection between two endpoints, or between two points that provide a partial link (e.g., a trunk) between two endpoints.

**Call Blocking** - a Quadro service that allows blocking unwanted incoming or outgoing calls over Quadro.

**Call Forwarding** - a Quadro service that allows transferring a call to another destination in case the Quadro user is busy, not answering or unconditional.

**Call Hold** - a Quadro service that allows holding the call in order to make another one, or to answer the second incoming call. The first call partner will listen to music while being on hold.

**Call Waiting** - a Quadro service that allows receiving a second call while being busy with the first one. The waiting party will hear a beeping during the conversation.

**Caller ID** - caller information is displayed on the called party's phone.

**Central Office (CO)** - a local switching system that connects lines to lines and lines to trunks. Sometimes used to refer to the building in which a switching system is located and the associated equipment. It is also the physical point where calls enter the long distance network.

**CODEC** - COmpression/DECompression that transforms analog voice into a digital bit stream and vice-versa. It is now an overall term for the technology used in digital audio and video.

### D

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**D-channel** - In ISDN, the 16-kb/s segment of a 144-kb/s, full-duplex subscriber service channel that is subdivided into 2B+D channels, i.e., into two 64-kb/s clear channels and one 16-kb/s channel for the ISDN basic rate. **Note 1:** The D channel is usually used for out-of-band signaling. The two 64-kb/s clear channels are used for subscriber voice and data services. **Note 2:** The D-channel specifications are addressed in the CCITT Recommendation for the Integrated Services Digital Network (ISDN). **Note 3:** The D-channel may be 64 kb/s for the primary rate ISDN service.

**Data Encryption Standard (DES)** - a block cipher algorithm for encrypting (coding) data so it is nearly impossible for anyone without the decryption key to get the data back in unscrambled form. The DES standard enciphers and decipheres data using a 64-bit key.

**Dial peer** - an addressable call endpoint. In Voice over IP (VoIP), there are two types of dial peers: POTS and VoIP.

**Dial plan** - a description of the dialing arrangements for customer use on a network.

**Digital Signal Processor (DSP)** - A specialized microprocessor that performs calculations on digitized signals that were originally analog, and then forwards the results. The big advantage of DSPs lies in their programmability. DSPs can be used to compress voice signals to as little as 4,800 bps. DSPs are an integral part of all voice processing systems and fax machines.

**Digital Subscriber Line (DSL)** - public network technology that delivers high bandwidth over conventional copper wiring at limited distances. There are four types of DSL: ADSL, HDSL, SDSL, and VDSL. All are provisioned via modem pairs, with one modem located at a central office and the other at the customer site. Since most DSL technologies do not use the entire bandwidth of the twisted pair, there remains room for a voice channel.

**Distinctive Ringing** - Quadro service that allows a specific ringing pattern assignment for particular callers over Quadro.

**Domain** - a place on the Internet you can visit with your browser, i.e., a www site. It also might be a single computer or computers masqueraded as a single computer. On the Internet, the domain is the address that gets you there.

**Domain name** - in a network using the TCP/IP, the full domain name consists of a sequence of names (labels) separated by periods (dots), for example, Quadro.epygi.com.



**Domain Name System (DNS)** - a system used on the Internet for translating names of network nodes into their addresses.

**Downstream** - in communications, there are two circuits. One coming toward you and the other going away from you. Downstream is another term for the transmission coming toward you.

**Dual-Tone Multifrequency (DTMF)** - a method of signaling consisting of a push-button or touch tone dial that sends out a sound consisting of two discrete tones that are picked up and interpreted by telephone switches (either PBXs or central offices).

**Dynamic Host Configuration Protocol (DHCP)** - a network standard regulating the IP address and other information assigned to the clients by the server.

**Dynamic Host Control Protocol (DHCP)** - a protocol that is used to dynamically allocate and assign IP addresses. DHCP allows you to move network devices from one subnet to another without administrative attention.

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## E

**E1** - wide area network digital transmission scheme. E1 is the European equivalent of a T1 line. The E1's higher clock rate (2.048 MHz) allows for 32 separate 64Kbps channels, which include one channel for framing and one channel for D-channel information.

**Ethernet** - a local area network used for connecting computers, printers, workstations, terminals, servers, etc., within the same building or campus. Ethernet operates over twisted pair and/or over coaxial cable at speed up to 10Mbps.

**Ethernet Controller** - the unit that connects a device to the Ethernet cable.

**Ethernet Switch** - the device that connects local area networks.

**Extensions** - users over Quadro.

**External User** - users connecting Quadro by IP or PSTN calls.

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## F

**Firewall** - a combination of hardware and software that limits the exposure of a computer or group of computers to an attack from outside. A firewall is a system or combination of systems that enforce a boundary between two or more networks. One purpose of an Internet firewall is to provide a single point of entry where a defense can be implemented, allowing access to the Internet resources from within the organization, and providing controlled access from the internet to hosts inside the organization's internal networks.

**Firmware** - is computer or OS required software that resides on ROM

**Foreign Exchange (FX)** - a Central Office trunk that has access to a distant Central Office. A dial tone is returned from that distant Central Office and a location can be reached in the area of the foreign Central Office by dialing a local number.

**Foreign Exchange Office (FXO)** - a service that can be ordered from the telephone company that provides local telephone service from a central office that is outside (foreign to) the subscriber's exchange area. To generate a call from the computer telephony system to the POTS set, you will need a FXS connection configured. See also FXS.

**Foreign Exchange Station (FXS)** - Interface that connects directly to a standard telephone, fax machine, or similar device over a standard RJ-11 modular telephone cable, and supplies ringing voltage, dial tone, and similar signals to it. see FXO

**Framing** - A procedure for controlling errors. Consists of inserting bits so the receiver can identify the time slots allocated to each subchannel

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## G

**Gatekeeper** - is the central control entity that performs management functions in a Voice and Fax over IP network and for multimedia applications such as video conferencing. Gatekeepers provide intelligence for the network, including address resolution, authorization, and authentication services, the logging of call detail records, and communications with network management systems. Gatekeepers also monitor the network for engineering purposes as well as real-time network management and load balancing, controlling bandwidth, and providing interfaces to existing legacy systems.

**Gateway** - an entrance into and out from a communications network. Technically, a gateway is an electronic repeater that intercepts and steers electrical signals from one network to another.

**Greeting** - voice messages that are played to the Quadro users or users calling to the Quadro activating specific services.

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## H

**Hold Music** - music played to the party that is on hold.

**Host** - an intelligent device attached to the network; can be also a mainframe computer.

**Host Name** - the name given to a mainframe computer or device.

**Hunt Grouping** - the Quadro service that allows configuring several users over Quadro to ring in series when a specific call arrives.

**Hypertext Transfer Protocol (HTTP)** - the protocol used by Web browsers and Web servers to transfer files, such as text and graphics files.

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## I

**Integrated Services Digital Network (ISDN)** - is a system of digital phone connections which allows voice and data to be transmitted simultaneously across the world using end-to-end digital connectivity. There are two basic types of ISDN service: Basic Rate Interface (BRI) and Primary Rate Interface (PRI). BRI is a basic service is intended to meet the needs of most individual users. PRI is intended for users with greater capacity requirements

**Internet Control Message Protocol (ICMP)** - a network-layer Internet protocol that reports errors and provides other information relevant to IP packet processing.

**Internet Protocol (IP)** - a unique, 32-bit number for a specific TCP/IP host on the Internet, normally printed in decimal form (for example, 128.122.40.227). Part of the TCP/IP family of protocols, it describes the software that tracks the Internet address of nodes, routes outgoing messages, and recognizes incoming messages.

**Internet Service Provider (ISP)** - a vendor who provides direct access to the Internet or a company that provides Internet access to other companies and individuals.

**Intrusion Detection System (IDS)** - is a firewall, but together with deleting the dangerous packets or packets including intrusion attacks, IDS also keeps information about dropped packets and the senders responsible for them.

**IP address** - also known as the Internet Address, is a unique 32-bit identifier for a specific TCP/IP host computer on a network. IP addresses are in dotted decimal form, such as 192.168.10.26, with each of the four address fields assigned as many as 255 values.

**IP address Mask** - A range of IP addresses defined so that only machines with IP addresses within the range are allowed access to an internet service. To mask a portion of the IP address, replace it with the asterisk wild card character (\*). For example, 192.44.\*.\* represents every computer on the internet with an IP address beginning with 192.44

**IP Gatekeeper** - defines the policies that govern a multimedia system such as dialing plans, user privileges, bandwidth consumption, and others. The gatekeeper also provides the means to extract information from such a system for various purposes, e.g., billing information, users that are logged in, etc. The gatekeeper is also a focal point for the introduction of supplementary services.

**IP Gateway** - most commonly, a network device that converts voice and fax calls, in real time, between the public switched telephone network (PSTN) and an IP network. The main IP gateway functions include voice, fax, compression/decompression, packetization, call routing, and control signaling. Additional features may include interfaces to external controllers, such as gatekeepers or soft-switches, billing systems, and network management systems.

**IP PBX** - an enterprise-based IP data network device that switches VoIP telephone traffic.

**IP Telephony** - a technology that allows voice phone calls to be made over the Internet or other packet networks using a PC via gateways and standard telephones.

**IPSec** - is used to provide security for transmission of sensitive information over unprotected networks such as the Internet. IPSec acts at the network layer, protecting and authenticating IP packets between participating IPSec devices ("peers"), such as Cisco routers.

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## J

**Jitter Buffer** - the buffer that collects incoming packets to place them in the right order. If the network has a high delay variation, increasing the Jitter Buffer can improve the audio quality, but this also increases the delay.

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## L

**LED** - Light-Emitting Diode, A semiconductor device that emits visible light when conducting current. Has replaced incandescent lamps as indicators in most electronic equipment.

**Lifeline POTS** - a voice telephone line that works even if electricity is cut off at the customer premises, since the line is powered from emergency backup at the central office. Multiple lifeline POTS lines can be delivered on one copper pair with the use of a digital line powered pair gain system. A basic telephone service supplying standard single line telephones, telephone lines, and access to the PSTN.

**Local Area Network (LAN)** - a short distance data communications network (typically within a building or campus) used to link computers and peripheral devices under some form of standard control.

**Login** -the procedure of identifying a user with a username and a password to enter into the protected field.

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## M

**Many Extensions Ringing** - a Quadro service that allows configuring several users over Quadro to ring simultaneously when a specific call arrives.

**Media Access Control (MAC) Address** - the address for a device as it is identified at the Media Access Control layer in the network architecture.

**Media Access Control (MAC) Layer** - is one of two sublayers that make up the Data Link Layer of the OSI model. The MAC layer is responsible for moving data packets to and from one Network Interface Card (NIC) to another across a shared channel.

**Media Gateway** - a generic class of products grouped under the Media Gateway Control Protocol (MGCP). A major function of the media gateway is simple IP/TDM conversion under the control of a softswitch.

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## N

**Name server** - a directory service that provides a mapping between a resource's global name and its physical location in the network.

**Network Address Translation (NAT)** - is used to allow LAN devices that do not have their own static IP addresses to connect to the Internet sharing an IP address. NAT will assume control of assigning their IP address. Furthermore, the NAT takes care that packets will reach the LAN PC that originated the traffic. This mechanism is absolutely transparent for the users (or the PCs in the LAN).

**Network Time Protocol (NTP)** - a protocol that is used for time counting in the Internet, based on the atomic clocks with the precision in milliseconds. This is the recommended protocol for synchronizing the time of hosts in the network.

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## P

**Packetization Interval** - the time interval between two RTP packets of the same stream. If the interval is increased, the overhead is decreased but the voice quality might deteriorate. If the interval is decreased, the network load is increased and the delay is reduced.

**Password** - a secret alphanumeric string used to identify and to allow the user to have access to a system.

**PCM** - a form of modulation in which the information signals are sampled at regular intervals and a series of pulses in coded form are transmitted representing the amplitude of the information signal at that time.

**Point-to-Point Protocol (PPP)** - allows a computer to connect to the Internet with a standard dial-up telephone line and a high-speed modem and to enjoy most of the benefits of the direct connection.

**Point-to-Point Tunneling Protocol (PPTP)** - enables virtual private networking - enabling secure remote access to corporate networks over the Internet.

**POTS (Plain Old Telephone Service)** - is the standard telephone service that most homes use. It is also referred to as the PSTN, or the Public Switched Telephone Network

**Private Branch Exchange (PBX)** - a telephone switch owned privately, usually by a large company. If it owns a PBX, a company does not need to lease a telephone line for each telephone set at a site.

**Proxy server** - an intermediate device that receives SIP requests from a client and then initiates requests on the client's behalf.

**Public Switched Telephone Network (PSTN)** - refers to the local telephone company.

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## R

**Real-Time Transport Protocol (RTP)** - the Internet-standard protocol for the transport of real-time data, including audio and video, allows applications to synchronize audio and video information. RTP connections are established between servers across the Internet after voice has been converted to IP format. RTP is used in virtually all Voice-over-IP architectures, for videoconferencing, media-on-demand, and other applications.

**Real-Time Transport Control Protocol (RTCP)** - is the control protocol that works in conjunction with RTP. RTCP control packets are periodically transmitted by each participant in an RTP session to all other participants. Feedback of information to the application can be used to control performance and for diagnostic purposes.

**Registration** - procedure of user subscribing to a server. Usually some personal parameters such as username, password, etc., are required upon registration.

**Remote Testing** - remote connection from the Epygi Support office to the customer's Quadro for testing and/or for troubleshooting.

**Router** - A device that determines the next network point to which a data packet should be forwarded enroute toward its destination. The router is connected to at least two networks and determines which way to send each data packet based on its current understanding of the state of the networks it is connected to. Routers create or maintain a table of the available routes and use this information to determine the best route for a given data packet

**RSA** - is an asymmetric key system. It must be available on both sides of the VPN and generates on each side a different pair of keys, a private and a public key.

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## S

**Security Parameter Index (SPI)** - is an index to keep VPN tunnels distinct. A security association is defined by destination, protocol and SPI. Without the SPI, connections to the same gateway using the same protocol would not be distinguishable.

**Session Initiation Protocol (SIP)** - is an application-layer control protocol that can establish, modify and terminate multimedia sessions or calls. SIP is increasingly used for Internet telephony signaling, in gateways, PC phones, softswitches, and softphones, but it is not limited to Internet telephony, and can be used to initiate and manage any type of session, including video, interactive games, and text chat.

**Signaling** - a process of sending a transmission signal over a physical medium for communication.

**Silence Suppression** - a method that allows disabling RTP packet transmission when there is no voice activity. This feature helps to avoid extra traffic when the RTP stream doesn't contains voice data.

**Simple Network Management Protocol (SNMP)** - the Internet standard protocol developed to manage nodes on an IP network.

**SIP address** - unique address of the users registered on the SIP server. The address can be used to connect the user. The full SIP address has the following format: "display name" <username@ipaddress:port>.

**SIP server** - this server is used for registering users. It gives a possibility to make IP connections between users registered on the same SIP server.

**Software** - PC programs.

**Software PBX** - a telephone system that converges voice and data on an industry-standard computing platform and uses computer telephony components that conform to industry standards. Since they conform to industry standards, software PBXs are interoperable with third-party systems and CT components. Conformance also allows software PBXs to run third-party enhanced applications such as desktop call control, graphical voice mail, automatic call distribution (ACD), IP gateways, follow-me call forwarding, unified messaging, and CRM integration.

**Speed Calling** - a service that allows making a personal address book for every Quadro user. A simple digit combination can be assigned to any destination phone number.

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## T

**Transfer** - a service giving a possibility to readdress incoming calls. Call Transfer can be conditional (with consultation) and unconditional (without consultation).

**Transmission Control Protocol (TCP)** - a connection-oriented transport layer protocol that provides reliable full-duplex data transmission. TCP is part of the TCP/IP protocol stack.

**Transmission Control Protocol/Internet Protocol (TCP/IP)** - is a networking protocol that provides communication across interconnected networks, between computers with diverse hardware architectures and various operating systems.

**Trunk** - is a communications channel between two points, typically referring to large-bandwidth telephone channels between switching centers that handle many simultaneous voice and data signals.

**Trunk Level 1 (T1)** - a high-speed (1.544Mb/s) digital telephone line with the equivalent of 24 individual 64Kb/s channels that are joined via time division multiplexing. A T1 line can be used to transmit voice or data, and many are used to provide connections to the Internet. T1 is the North American equivalent of an E1 line.

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## U

**UDP** - a connectionless transport layer protocol in the TCP/IP protocol stack. UDP is a simple protocol that exchanges datagram without acknowledgments or guaranteed delivery, requiring that error processing and retransmission be handled by other protocols.

**Universal Serial Bus (USB)** - is an interface with a protocol that is designed to handle a broad range of devices - telephones, modems, printers, etc.

**Upstream** - in communications, there are two circuits - one coming toward you and the other going away from you. Upstream is another term for the name of the channel going away from you.

**URL** - an identifier used to locate content that is transported via the HTTP protocol.

**Username** - identification name of the user. Usually used for registration and login.

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## V

**VCI** - parameter used to configure ATM settings and is usually given by the Internet provider.

**Virtual Private Network (VPN)** - connects two local networks (intranets) over the insecure Internet securely. VPN routers manage authentication between servers and clients and handle data encryption for the connection. Only authorized users can access the network and the data exchange cannot be intercepted. A VPN includes authentication and encryption to protect data integrity and confidentiality. VPNs are "virtual" in the sense that individuals can use the public Internet as a means of securely accessing an internal network. Once the VPN connection is established, users have access to the same network resources, addresses, and so forth as if they were connected locally. VPNs are "private" because the data is encrypted between two VPN gateways. Encryption makes it very difficult for anyone to intercept data and capture sensitive information such as passwords.

**Voice mail** - a brief message that external users can leave for the Quadro users in the event that nobody answers the call.

**Voice Mail System (VMS)** - a feature providing the possibility of leaving brief voice messages at the unavailable or busy Quadro extension's mailbox.

**Voice mailbox** - is the mailbox where voice mails are collected.

**Voice message** - help messages that are played to the user giving a hint on how to manipulate the menus within Quadro using the phone handset.

**Voice Over Internet Protocol (VOIP)** - technology used to transmit voice conversations over a data network using the Internet Protocol. The ability to carry normal telephony-style voice over an IP-based internet with POTS-like functionality, reliability, and voice quality.

**VPI** - parameter used to configure ATM settings usually given by the Internet provider.

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## W

**Wide Area Network (WAN)** - a communications network used to connect computers and other devices across a large area.