

Configuration Last Call Repeat Facility



Administration manual for system providers

11/19/2020

Product line neo, version 6.x

The described functions can be used with the following ASC products:

EVOIPneo

EVOLUTIONneo / XXL / eco

INSPIRATIONneo

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Contents

1	General information	4
2	Introduction	5
3	System requirements.....	6
4	Licenses	7
5	Configure Last Call Repeat Facility	8
5.1	Tab Usage.....	8
5.2	Tab Media Streamer	9
5.3	Configure language for announcements	11
5.4	Configure announcement texts	13
5.5	Create new PBX.....	14
	Glossary	17

1 General information

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2 Introduction

This manual describes how to start up the feature *Last Call Repeat Facility*.

The feature *Last Call Repeat* allows accessing calls by phone which have been recorded by a neo recording system.

The recorded audio data can be replayed by means of an analog or a digital or a VoIP end device.

Up to 4 users can call the [replay server](#) from the telephone network at the same time. Upon authenticating by means of the user name and a password, users can replay the latest calls that have been recorded.



For information about the usage refer to the user manual *Last Call Repeat Facility*.

The called replay server can replay recordings by means of the integrated *Replay Feature*. Only data which has either been recorded directly on the replay server or which has been transferred to the replay server for data storage or only for replay purposes can be replayed.

If no SIP interface is available to connect the recording server to the PBX, an external analog gateway is required.

The *Last Call Repeat Facility* is configured in the Servers module of the application System Configuration.



Basic information about using the application System Configuration can be found in the user manual for administrators *System Configuration - General information*.

3 System requirements

3 System requirements

To use the feature, note the following:

If no [SIP](#) interface is available to integrate the neo recording server into the PBX, an external analog gateway is required (included in the delivery of the license *Last Call Repeat*).

4 Licenses

ASC

License name	Number	Description
<i>Last Call Repeat</i>	1	License required for using the <i>Last Call Repeat Facility</i> , available for 4 or 8 concurrent replays.

Tab. 1: Licenses of ASC

5 Configure Last Call Repeat Facility

The feature Last Call Repeat has to be configured in the application System Configuration in the Servers module.

1. Open the menu item *Setup > Servers* in the navigation bar.
2. In the main view, select the server that you would like to configure the feature for.

5.1 Tab Usage

1. Click on the tab *Usage* in the detail view.
2. Open the group field *Replay*.

In this tab, you can configure the purpose of the selected server.

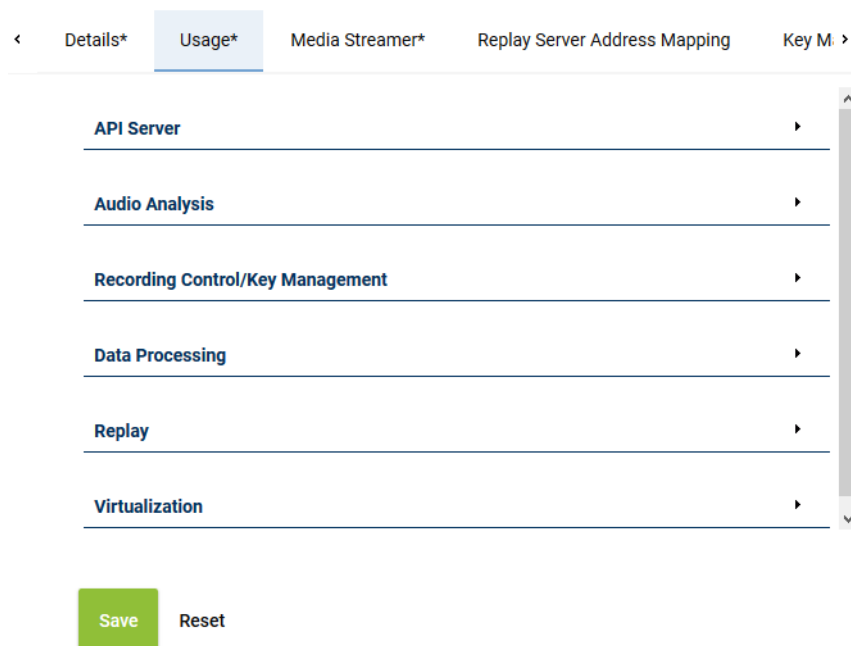


Fig. 1: Servers module - tab Usage

Group field Replay

1. Open the group field *Replay*.

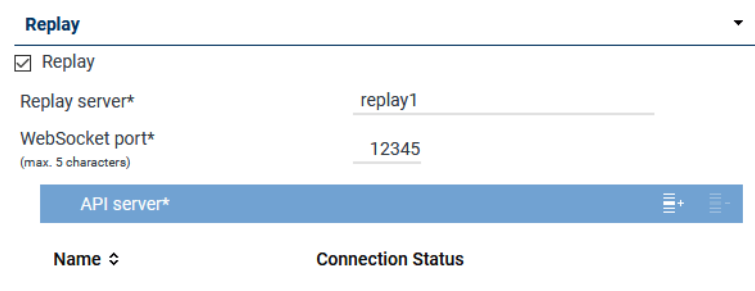




Fig. 2: Group field Replay

Replay

Select whether the server is supposed to serve as replay server.

A replay server can replay recordings via the integrated *Replay Feature*. Only data which has either been recorded directly on this server or which has been transferred to this server for data storage or only for replay purposes can be replayed. The client computers of the system can connect to a replay server for replay purposes.

	<input checked="" type="checkbox"/> = Function has been activated. You have to complete the entry field <i>Replay server</i> . <input type="checkbox"/> = Function has not been activated.
<i>Replay server</i>	<p>Enter the name which is supposed to denote the server as the replay server in the system.</p> <p>As the replay server can be used system-wide and by different tenants, you have to enter a kind of alias here. When selecting the replay server, this alias is displayed on the client computers instead of the real server name or the IP address.</p>
<i>WebSocket port (maximum of 5 characters)</i>	<p>Enter the port via which the data to be replayed in POWERplay Web are supposed to be transmitted.</p> <p>In order to be able to reach the replay server from a public network and with configured port forwarding, you have to adjust the settings in the tab <i>Applet Address Mapping</i>.</p> <p>Keep in mind that the indicated port must have been opened.</p>
List <i>API server</i>	<p>Here, you can add API servers for replay. If a recording which is supposed to be replayed cannot be found on the server, the search is continued on the storage expansions which have been entered here. That way, even recordings can be replayed which have not been transferred to the server.</p> <p>If the function <i>Replay</i> has been activated, you can adjust the following settings:</p> <ul style="list-style-type: none"> • By clicking on the icon  (<i>Add</i>) you can add the API server. • By clicking on the icon  (<i>Remove</i>), you can remove selected API servers from the list.

- To save the entries, click on the button *Save*.
To discard entries, click on the button *Cancel*.

5.2

Tab Media Streamer

- Click on the tab *Media Streamer* in the detail view.



The tab *Media Streamer* is only active if the function *Replay via phone* has been activated in the tab *Usage*.

In this tab, you can configure the Media Streamer for the functionalities *Replay via phone* and *Last Call Repeat Facility*.

<
Details*
Usage*
Media Streamer*
Replay Server Address Mapping
Key M. >

PBX +

PBX	PBX	▼
Extension* (max. 18 characters)	123456	
Media streamer IP address*	192.168.169.192	▼
Minimum port	24000	
Maximum port	24099	
Transport protocol	UDP	▼
SIP signaling port	5062	
User name		
Password		
PBX IP address		
PBX port	5060	
Registration required	<input checked="" type="checkbox"/>	
SIP registration expiration	3600	Second(s)

Save
Reset

Fig. 3: Servers module - tab Media Streamer

2. Enter the following parameters:

PBX	<p>PBX that the Media Streamer is supposed to be mapped to.</p> <p>Select a PBX from the drop-down list. The drop-down list displays all PBXs which have been created in the system.</p> <p>If no PBX has been created in the system yet, you can create a PBX via the blue bar PBX, see Create PBX.</p>
Extension	<p>Extension which is supposed to be mapped to the Media Streamer. This is a mandatory field; the configuration cannot be saved if this information is missing.</p> <p>If an external analog gateway has been integrated, enter the value 8000.</p>
Media streamer IP address	<p>IP address which is supposed to be used for the exchange of the audio data and for the SIP communication.</p> <p>Select an IP address from the drop-down list. In the drop-down list, all IP addresses of the server are displayed.</p> <p>If an external analog gateway has been integrated, select the IP address 169.254.254.100 in the drop-down list.</p>
Minimum port	<p>Enter the minimum port which is supposed to be used for the audio data exchange.</p>
Maximum port	<p>Enter the maximum port which is supposed to be used for the audio data exchange.</p> <p>A port range of 100 (e. g. 24000-24099) is sufficient for 50 licenses. The port range should be twice as wide as the number of available licenses.</p>
Transport protocol	<p>Select the transport protocol type you would like to use for the SIP communication from the drop-down list.</p>

	<p>TCP = unencrypted</p> <p>UDP = unencrypted</p> <p>TLS = encrypted</p> <p>If an external analog gateway has been integrated, select <i>UDP</i> in the drop-down list.</p>
<i>SIP signaling port</i>	<p>Enter the port for the <i>SIP</i> communication.</p> <p>Port for data exchange: 5062</p>
<i>User name</i>	Enter the user name for the authentication on the <i>SIP</i> server.
<i>Password</i>	Enter the password for the authentication on the <i>SIP</i> server.
<i>PBX IP address</i>	<p>Enter the IP address of the <i>SIP</i> registrar of the <i>PBX</i>.</p> <p>If an external analog gateway has been integrated, enter the IP address 169.254.254.101.</p>
<i>PBX port</i>	<p>Enter the port of the <i>SIP</i> registrar of the <i>PBX</i>.</p> <p>If an external analog gateway has been integrated, enter the value 5060.</p>
<i>Registration required</i>	<p>Select whether the <i>SIP</i> extension has to be registered with the <i>SIP</i> registrar of the <i>PBX</i>.</p> <p><input checked="" type="checkbox"/> = <i>SIP</i> extension has to be registered.</p> <p><input type="checkbox"/> = <i>SIP</i> extension does not have to be registered.</p> <p>If an external analog gateway has been integrated, deactivate the check box <i>Registration required</i>.</p>
<i>SIP registration expiration</i>	Enter the time interval after which the registration has to be repeated.

- To save the entries, click on the button *Save*.
To discard entries, click on the button *Cancel*.

5.3 Configure language for announcements



Make sure that the function *Replay via phone* has been configured and that at least 1 conversation is replayed via phone. Upon replaying a conversation via phone for the first time, the file *ASC.LocalReplayService.ini* is created to configure the language.



The entered language is valid across the system for all tenants.

To change the language for announcements in *Replay via phone*, proceed as follows:

- Log in on the replay server with administrator rights.
- Open the Windows Explorer.
- Go to the directory *C:\Program Files (x86)\ASC\ASC Product Suite\data\LocalReplayService*.
- Right-click on the file *ASC.LocalReplayService.ini*.
⇒ The following context menu appears:

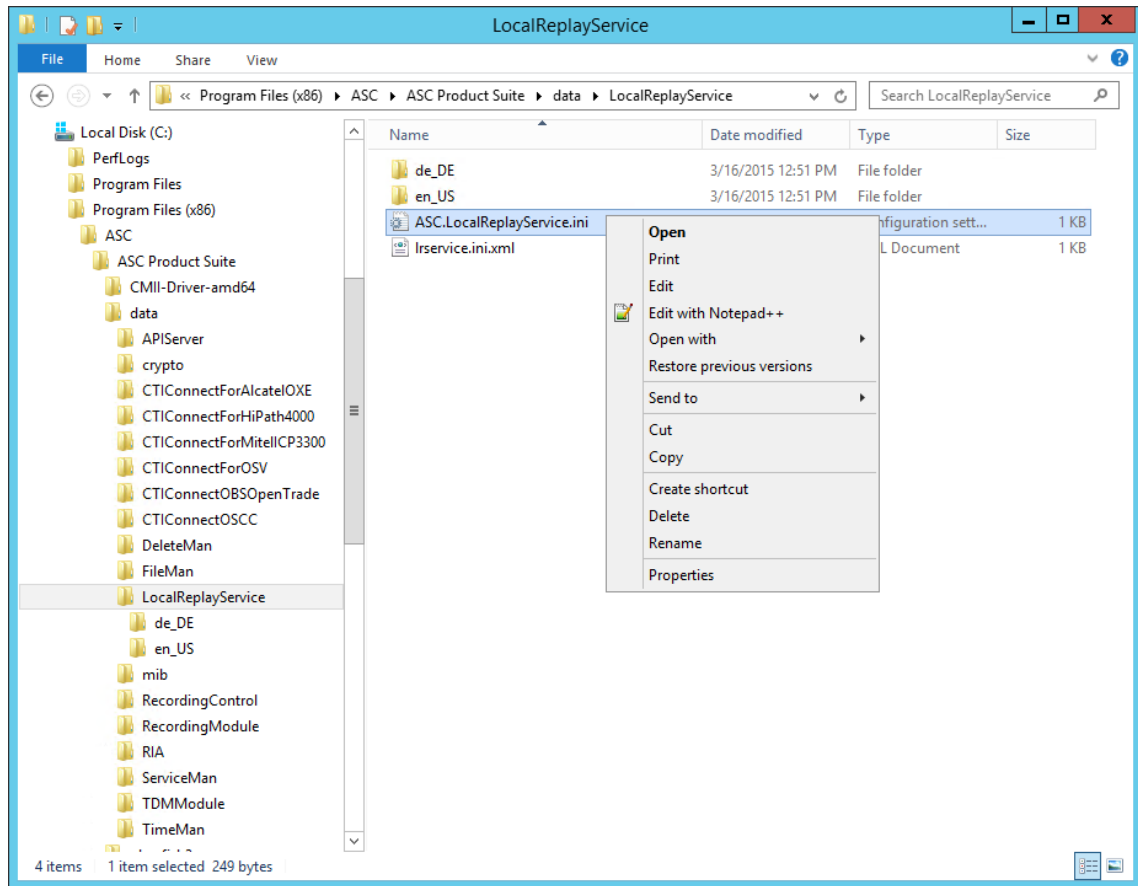


Fig. 4: Context menu for the file ASC.LocalReplayService.ini

5. Click on the menu item *Edit* in the context menu.

⇒ The following window appears:

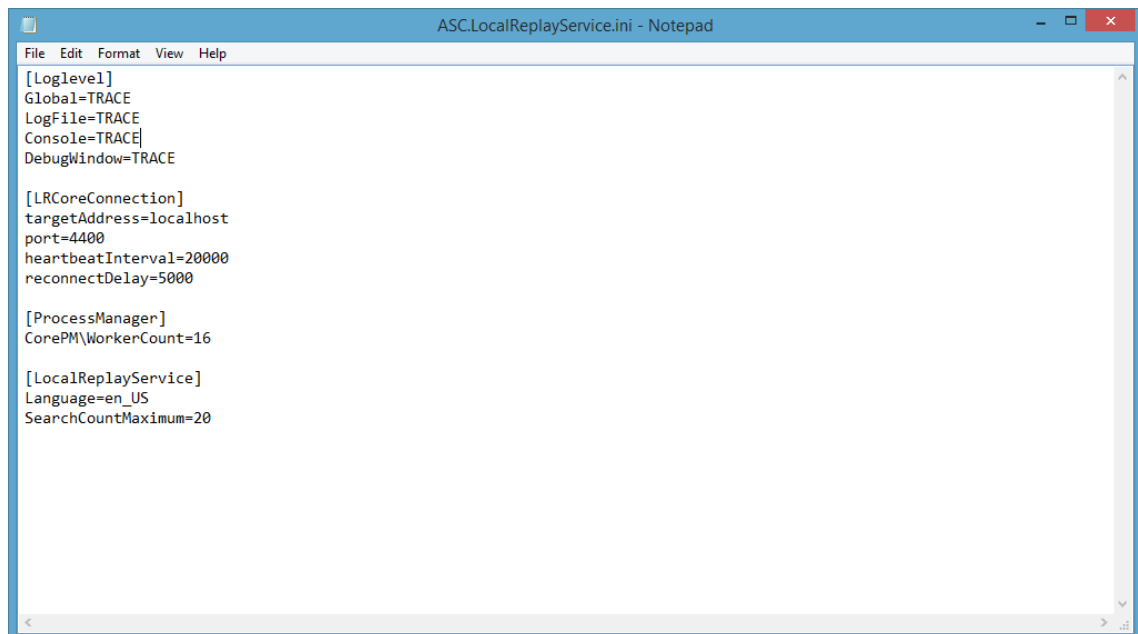



Fig. 5: Edit ASC.LocalReplayService.ini

6. In the section *[LocalReplayService]*, change the value for the parameter *Language*. One of the following values can be entered:
 - *Language=de_DE*
 - *Language=en_US*

7. In the menu bar, click on *File > Save*, to save the changes.
8. Click on the icon  (*Close*) to close the window.

5.4 Configure announcement texts

The announcement texts have been saved as .PCM files and can be replaced by customized announcement texts.

The call data of the LCR server has the following format:

- 8 bit, 8 KHz, PCM [μ-law](#)

The call files have been saved in the following directories according to their language:

German:

- *C:\Program Files (x86)\ASC\ASC Product Suite\data\LocalReplayService\de_DE*

English:


- *C:\Program Files (x86)\ASC\ASC Product Suite\data\LocalReplayService\en_US*

The following table shows the default call files:

File name	Announcement text
<i>Begin.pcm</i>	You are at the beginning of the call.
<i>CheckOut.pcm</i>	You are logged off. Goodbye.
<i>ConnectErr.pcm</i>	We are unable to establish a connection to the server. Please try again later.
<i>End.pcm</i>	You are at the end of the call.
<i>Leave.pcm</i>	Thank you for your call. Goodbye.
<i>LicenseErr.pcm</i>	There are no more licenses available.
<i>LoginErr.pcm</i>	Your login has failed. Please enter your user ID and confirm by pressing the star key. To end the call, press the pound key. If you are still unable to log in, please contact your system administrator.
<i>MaxHookOff.pcm</i>	The time limit for the call has been reached.
<i>MaxLoginTries.pcm</i>	We are still unable to recognize your login. Please contact your system administrator.
<i>Menue.pcm</i>	To start playback, press 1. For rewind, press 2. For fast forward, press 3. To stop playback, press 4. For the previous call, press 5. For the next call, press 6. To pause, press 7. To logoff, press the star key. To end the call, press 9.
<i>n_a.pcm</i>	There is no audio data available for this call.
<i>Nofunction.pcm</i>	You have pressed an invalid key.
<i>NoNextCall.pcm</i>	That was the last call.
<i>NoPreviousCall.pcm</i>	This is the first call.
<i>Rcrid.pcm</i>	Please enter your password and confirm by pressing the star key. To end the call, press the pound key.
<i>Rcruser.pcm</i>	Please enter your user ID and confirm by pressing the star key. To end the call, press the pound key.
<i>SearchErr.pcm</i>	There are no calls that match your request.
<i>WEBINTRO.pcm</i>	Hello. You are connected to the ASC voice recording system.
<i>Welcome.pcm</i>	Hello. You are connected to the ASC voice recording system. Please enter your user ID and confirm by pressing the star key. To end the call, press the pound key.

5.5

Create new PBX

1. In the blue bar with the title *PBX*, click on the icon  (*Create*).
2. Enter all necessary information.

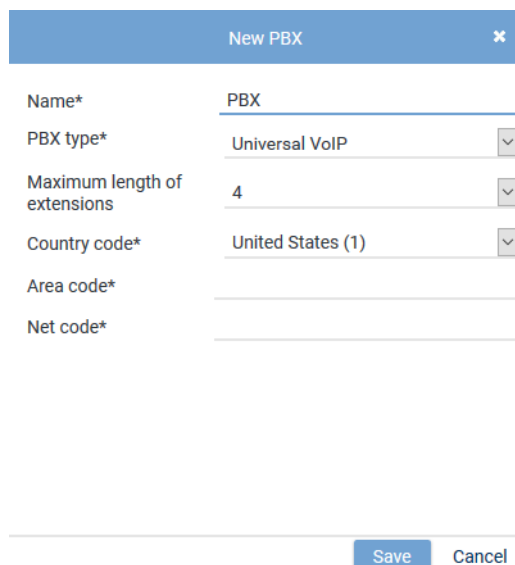


Fig. 6: Create PBX - example

<i>Name</i>	Enter a name for the new PBX instance.
<i>PBX type</i>	Select the PBX type from the drop-down list.
<i>Maximum length of the extensions</i>	Select the maximum length of the extensions from the drop-down list.
<i>Country code</i>	Select the country code from the drop-down list.
<i>Area code</i>	Enter the area code for the PBX .
<i>Net code</i>	Enter the net code for the PBX .

3. To save the settings, click on the button *Save*.
To discard the settings and close the window, click on the button *Cancel*.

List of figures

Fig. 1	Servers module - tab Usage	8
Fig. 2	Group field Replay	8
Fig. 3	Servers module - tab Media Streamer	10
Fig. 4	Context menu for the file ASC.LocalReplayService.ini	12
Fig. 5	Edit ASC.LocalReplayService.ini	12
Fig. 6	Create PBX - example	14



List of tables

Tab. 1 Licenses of ASC..... 7

Glossary

μ-law

PCM digitization method for analog audio signals according to ITU G.711. In the process, analog voice signals are converted into digital signals by means of a logarithmic quantization characteristic. The μ-law algorithm is used in the US while the A-law algorithm is the standard in Europe.

PBX

Private Branch Exchange

Replay server

Server on which the replay function has been activated. Recordings can be replayed via this server.

SIP

Session Initiation Protocol

TCP

Transmission Control Protocol, controlled connection establishment, secure data transmission, controlled connection termination

TLS

Transport Layer Security; previously known as Secure Sockets Layer (SSL), is a hybrid encryption protocol for safe data transmission in the Internet. Since version 3.0, the SSL protocol is developed under the new name TLS.

UDP

User Datagram Protocol UDP is a minimal, connectionless network protocol which belongs to the core members of the Internet protocol suite. Its purpose is to make sure that data transmitted via the Internet reach the designated application. There is no destination check.

VoIP

Voice over IP