Wildix

> ⓘ The List of custom config parameters that allow to change various settings in config files.
>
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>
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>
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Users

2 PBXs in WMS Network, each with its own Active Directory for users

> ⚠ You need admin access to Active Directory server.

To make it work, proceed as follows:

1. Make import of users via Active Directory on Server PBX
2. Access Client PBX and move users from Server PBX to Client PBX
3. Enable Active Directory sync on Client PBX: connect as root via SSH to Client PBX and create the file /rw2/etc/ad_connect.conf
4. Copy the contents of the file ad_connect.conf from Server PBX to Client PBX

Result: Single Sign-On for Active Directory works for users on Client PBX.

> ⚠ Limitation: the option "Remove existing users which are not received from the backend" does not work in this configuration; do not enable it.

Devices

Disable devices sync

> ⊘ Important: By default, the sync is enabled. It is necessary to disable it in case of FAILOVER scenario on Failover PBXs!

To disable the sync:

- Add parameters *disable_sync_portal=1* " and *disable_sync_manual=1* to the config file /rw2/etc/pbx/device_sync.conf
  Available values: 1 – sync is disabled; 0 – sync is enabled

Enable g729 transcoding for web phone calls to trunks which do not support G711

> ⊘ Important:
>
> - It's not recommended to enable this feature, as it reduces call quality and generates useless load on CPU!
> - It must be enabled only if the operator doesn't support g711a/u for some calls
> - It can generate CPU overload and problems if too many calls use it; in this case it is recommended to use another operator which supports all the needed codecs (g711a / g711u / g729)

> ⚠ Note: Feature is supported only on PBXs with modern CPU or Cloud.

To enable g729 transcoding, edit the file /etc/callweaver/sip-general-custom.conf:

- Add the parameter *g729_transcoding=yes*

Enable HD codecs on PBX (Beta version, WMS 4.01)

> ⚠ Notes:
>
> - Currently supported devices: Collaboration, Android v. 4.01.12/ iOS v. 7.1.35284 apps, WelcomeConsole v. 68.145.2.1
> - The feature works for PBXs in WMS Network

The feature is disabled by default. To enable it:

- Edit the file /etc/callweaver/sip-general-custom.conf by adding the following lines:

```
disallow=all
allow=opus:20,g729:20,alaw:20,ulaw:20,vp8,h264,h263,h263p
```

- Run the command:

```
callweaver -rx "sip reload"
```

Trunks

Enable Q-value (serial forking) for trunk registration

To enable q-value (serial forking) parameter via custom register string:

- Copy registration line for some trunk from /etc/callweaver/sip-registration.conf (Example: register => 144?144:123456:"144@10.168.0.144:5060)
- Uncheck *Enable registration* option in Trunk Settings (WMS -> *Trunks*)
- Add new line into /etc/callweaver/sip-general-custom.conf: register => 144?144:123456:"144@10. 168.0.144:5060~~0.6 (where 0.6 is q-value)
- Run the command:

```
callweaver -rx "sip reload"
```

Enable sending keep-alive packets via UDP packets to keep RTP ports opened

The option improves symmetric RTP/ NAT by allowing keep-alive packets to be sent via UDP packets for PBXs located behind NAT.

To send UDP packets (by default, RTP packets are sent), proceed with the following:

- Add the parameter *rtpkeepalive_mode=udp* (available values [udp|rtp]) to the file etc/callweaver/sip-general-custom.conf

Dialplan

Enable exit code 0 from voicemail

Starting from WMS Stable Version 3.88.41762.32, support for exit code 0 from voicemail allowing caller to speak with an operator was added

How to use:

- Add the letter 'o' as called number to the Dialplan context (that is where the "0" key sends the caller)

How to enable:

- Add the parameter *operator=yes* to the file voicemail.conf. It allows sender to hit 0 before/ after/ during leaving a voicemail to reach an operator

Allow overriding of Global Call groups settings

The option allows overriding of Global Call groups settings (see the chapter below) and saving custom parameters after each system upgrade.

To override Global Call group settings, you need to edit *queues-general-custom.conf* which is included in file *queues.conf*.

- Access *queues-general-custom.conf* via SSH as root:

```
nano-tiny /rw2/etc/callweaver/queues-general-custom.conf
```

- Add a custom parameter, for example, *autofill = yes* (by default, the file *queues.conf* contains *autofill = no* parameter)
- Save changes

Global Call groups settings

Global Call group settings are defined and configured in [*general*] section of the configuration file *queues.conf* (the path to the file: /rw2/etc/callweaver/queues.conf).

[general] section

The section contains global settings that are applied to all Call groups.

- *persistentmembers = yes*

With *persistentmembers* enabled, all dynamically added Call group members (via Feature code "Call group management" 97, WebAPI "Call group login" and Contact center feature in Collaboration) are stored in their Call groups and therefore saved after the system restart.

- *autofill = no*

With autofill disabled, a Call group attempts to deliver calls to members in a serial manner. This means only one call is attempted to be distributed to members at a time. Additional callers are not distributed to members until that caller is connected to a member. With autofill enabled, callers are distributed to available agents simultaneously. The parameter allows you to more efficiently distribute calls between Call group members, especially if there are several callers in a queue and several members can accept a call.

> ⚠ General information regarding Call groups: WMS Start Guide
>
> How to set up call distribution in Call groups: Call distribution in Call groups Admin Guide

Integrations

Hotel PMS

WHoteld package supports some custom configuration parameters that can be changed by editing the file /etc/wildix/whoteld_manager_custom.conf.

List of the available parameters:

- *ReSyncType* is used to modify data synchronization procedure. The following values are available:
    - *Full* - allows all synchronizations
    - *Forbid* - denies any data synchronizations
    - *Lite* - denies synchronization requests to FIAS/ XOpen interfaces
- *DnDBehaviour* - setup for the DND event processing. The following values are available:
    - *std* - accepts DND events only from FIAS interface
    - *extended* - allows handling DND events also from PBX and XOpen interfaces
- (removed to another file. See information below) *BadgeTimeout* - timeout for waiting on the badge programming response
    - Supported values: 2 min - 6 min - 10 min (min - default - max)

> ⚠ Note: Starting from WMS Beta Version 3.88.42955.49, the parameter "BadgeTimeout" is removed from *whoteld_manager_custom.conf* to whoteld_manager_custom.conf.
>
> The new default timeout is 60 seconds. New supported values: 20-600 seconds (min-max).

> ⚠ Note: Information about hotel integration: Hotel Integration - FIAS protocol - Guide.

Fax Server

Adjust the resolution of outgoing faxes

> ⚠ Note: The default resolution is 204x196 (fine).

To adjust the resolution:

- Edit the config file /rw2/etc/faxglobal.conf by specifying the desired horizontal and vertical resolution of the image in pixels per inch
- For example: XResolution=204 YResolution=391
- Other widespread resolutions: 204x98, 204x391, 408x391