

Extensions

This FAQ page collects operational notes for **Extensions**.

Can I use an extension to connect a remote PBX to the system?

Yes, but you need to enable the “trunk” feature for the extension or otherwise the Caller ID of the call coming from the remote PBX will be overwritten.

When I use server side attended or unattended transfer (#* and ##), there is no enough time to dial the destination extension!

The default timeout is set to 3 seconds, but the “transfer” message is played inside this time, so it may seem shorter. You can increase the timeout of the transfer by editing the `/etc/asterisk/features.conf` and changing the value for `transferdigittimeout` to the amount of seconds you like. Once done, reload the module from within asterisk with “`module reload features`”

When I try to recover the Voicemail, it says the PIN is invalid

Most of the time, there is a problem with DTMF, check the log for the call in `/var/log/asterisk/full` if you see this message: `dsp.c: Inband DTMF is not supported on codec g729. Use RFC2833` In this case, change the DTMF setting on the PBX from “auto” to RFC2833 and if possible, also on the phone

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