Pre-requisites and specification

1. Introduction

Each PrivateServer can reach outer communication devices using its Trunk feature. This means that it's possible to connect PrivateServer to almost anything out there, if that device talks standards protocols as SIP. The outer communication design can be:

- FROM the PrivateServer TO the other PBX/Device
- TO the PrivateServer FROM the other PBX/Device

In the former case we are going to use the "register" feature, in the latter we need less standard configuration. For what it was just written, it's common understanding that out first choice of interest is the first case, thus we configure PrivateServer to act AS A CLIENT and set it up to connect to another PBX /Device via CLEAN SIP over UDP TRUNK.

2. Pre-requisites

There is some pre-requisite to be filled in before we even start configuring. The most important one is the network connection, but we also got to take care of design, deciding either the number plan and the possible dial prefix.

2.1 Network

It can sound awful or just pedantic, but it's necessary that both the devices (ie PrivateServer and the other one) can "see" each other, and not just in matters of ICMP Pings, but actually via UDP connection. This because the UDP connection will be the transport mean for the SIP Trunk. Thus please check out the UDP communication is possible on the network legs between the two devices. We suggest to let the whole UDP port range open.

2.1 Number Plan

Also this pre-requisite is often disattended or considered of less importance. This attitude is wrong, because without a clear and stated number plan we can really risk number overlapping which would lead to unexpected call routings and very nasty to debug issues. So please before you start just write down the telephone numbers you are providing to the PrivateServer device to be called and consider the impact of calling secure numbers on your own PBX.

2.2 Dial Prefix

To let either PrivateServer and the other device correctly route the calls to each other, we need to decide whether to setup a dial prefix or not. Consider that the behaviour for PrivateServer is to search in its own database for the number dialed. If the number is not present AND a SIP Trunk is on, then PrivateServer will route the call throught the SIP TRUNK, expecting that on the opposite side the other device would know how to deal with it. So if you mean to route specific numbers (let's say actual phone numbers) from a secure leg to your PBX and throught the PrivateServer, you better consider to add a routing dial prefix to make them acknowledged.

3. Specification

After you're done with pre-requisites, then you can start configuring the devices. These are the protocol specifications you need to be implemented and set up to make the devices talk each other.

3.1 SIP protocol support

The SIP Trunk would work on... SIP protocol. So it's mandatory that such protocol is supported by the PBX/device to which PrivateServer is connecting.

3.2 Registration/non registration

It doesn't matter if the device supports the registration or doesn't, as PrivateServer can be configured in either ways, but this is a feature to be noted in order to correctly configure the trunk on PrivateServer

3.3 Codecs

The device must support the **alaw/ulaw** audio codec in order to be able to connect to PrivateServer.