PSAM 0. Introduction

The present manual offers a guide for installation and first time configuration of **PrivateServer** which along with **PrivateGSM** composes the **Enterprise Voice Security Suite**: the solution that ensures the safety and inviolability of voice communications on both mobile and fixed phones by offering a secure voice convergence system in a corporate network.

Requirements

PrivateServer works on VoIP technology (Voice over IP) and requires an internet access. Thus the server chosen for installation (either phisical or virtual) must have almost one Network Interface Card on board.

Goals

The present manual will explain to you how to:

- install **PrivateServer** from cdrom or by virtual machine
- configure the Network
- configure your certificates
- organize the network security architecture
- perform backup and restore
- perform software upgrades

Intended audience

This guide is intended for system administrators who will configure a secure enterprise voice network based on PrivateServer. The reader should have a networking and system administration background. VoIP knowledge is not mandatory, but strongly suggested.

This manual won't explain how to manage the service itself (ie create users, groups, SIP trunk, etc). For such knowledge please refer to the **PrivateServer User Manual**.

Glossary

In the present documentation you'll meet the following terms:

PBX

A **private branch exchange** (PBX) is a telephone exchange that serves a particular business or office, as opposed to one that a common carrier or telephone company operates for many businesses or for the general public.

PBXs make connections among the internal telephones of a private organization—usually a business—and also connect them to the public switched telephone network (**PSTN**) via **trunk** lines.

Trunk

A **trunk line** is a circuit connecting telephone switchboards (or other switching equipment), as distinguished from local loop circuit which extends from telephone exchange switching equipment to individual telephones or information origination/termination equipment.

When dealing with a **private branch exchange** (**PBX**), trunk lines are the phone lines coming into the **PBX** from the telephone provider. This differentiates these incoming lines from **extension** lines that connect the **PBX** to (usually) individual phone sets.

Extensions

A **telephone extension** is an internal telephone line attached to a Private branch exchange (PBX). The PBX operates much as a community switchboard does for a geographic telephone numbering plan and allows multiple lines inside the office to connect without each phone requiring a separate outside line. In these systems, a dialer usually has to dial a number to tell the PBX to connect with a landline to dial an external number. Within the PBX, the user merely dials the extension number of the person. Each phone line may be extended up to a fixed maximum.

Secure Call

A **Secure Call** is a voice connection which can't be wiretapped. It runs on Voice Over IP (VoIP) communication protocol and can operate in two security models:

- End to end
- End to site

The **end to end** provides the **highest** security level but can be used only between two PrivateGSM equipped devices.
The **end to site** provides a **strong** security level and can be used among two or more PrivateGSM equipped devices and/or among **SNOM 300** landline devices or also for connecting other **PBX,** secure or not. If PBX is not secure, we face a crypto-to-clear scenario, where the call is secured between **PrivateGSM** and **PrivateServer,** but is not secured between **PrivateServer** and **PBX**.

PrivateServer

PrivateServer is the **PBX** committed to perform **Secure Calls** both **end to end** and **end to site**. It differs from a standard **PBX** for exposing just the **Secure Call** service to VoIP **PrivateGSM** clients and can be connected to a standard **PBX** via **SIP Trunks** if configured accordingly.

PrivateGSM

PrivateGSM is the **VoIP client for Secure Calls** connections. It has to be used along with **PrivateServer** and is deployed in two different models to perform two different security models:

- **PrivateGSM Professional**: used for **end to end** communications
- **PrivateGSM Enterprise**: used for **end to site** communications

Because of the security model each one implements, the two versions can't communicate with each other. Plus, the **PrivateGSM Professional** edition can only perform direct calls to another PrivateGSM Professional Devices. On the other hand the **PrivateGSM Enterprise** Edition can also perform **Conference Calls** and **Three-Way Calls**.

Conference Calls

The **conference call** is a call in which more than one person can partecipate. The conference calls are usually defined as "**rooms**", whose access can be limited by **time settings** or **pass code**. In **PrivateServer** the **conference calls** can be provided on a **Secure Call** base if using the **PrivateGSM Enterprise** Edition only.

Three-Way Calls

The **Three-Way** call, as the name implies, is a call among **three partecipants**. The third user is added to a precedent running conversation by one of the two parteipants.

Call transfer

Call transfer is a typical PBX performance which is implemented in **PrivateServer** as well, but only if the users are running a **PrivateGSM Enterprise** Edition. One of the partecipant can hold on his/hers peer and perform a new call to the number to which trasfer the call. If the desired number picks up the call, then the transferrer can close the communication and let his/hers peer talk with the trasferred number.

Jitter

In VoIP systems audio signal is split into multiple packets, which are sent over network. Due to network equipment behavior, packets flow is never regular and constant. Especially on mobile/radio networks packets are delivered in bursts, leading to irregular and variable latency. **Jitter** is the variation in latency as measured in the variability over time of the packet latency across a network.

Automatic Activation

The **Automatic Activation** is the way of create new users automatically without any need they interact with anybody. As from user's side the procedure is that he/she gets first an **invite SMS** useful to download the application PrivateGSM and then a **configuration SMS** which provides automatic configuration of the client itself. All the user has to do is follow the links into both the SMSes and it goes automatically on line.

Provisioning

The **Provisioning** is the configuration needed for delivering both for the PrivateGSM application and its configuration and nowadays it's used by the **Automatic Activation** only.

Presence

The **Presence** is how we call the user's status, also known as the user's reachability. By checking an user's Presence it is possible to know if a he/she is on line and can receive a secure call before trying to.

Audio Messaging

The **Audio Messaging** is the means used by PrivateGSM for communicating to an user about the failed calls. You have several messages that can be spoken and each of them can be localised in English, French, Italian, Spanish and German.

Caveat

The figures in this document are solely for illustrative purposes. They give you an idea about the essential information you are supposed to see on the screen while executing the test cases. However the layout of the screen and the details of the information may be changed in subsequent revisions of the software and these modifications are not obligatory reflected in this document. When considering whether a test case passed or not, you should relay only on the textual description of the test case.

PSAM 1. Installation