PSUM 0 - Introduction

The present manual offers a guide to manage the **PrivateServer** that (among **PrivateGSM** clients) 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

An installed and on-line **PrivateServer**.

Goals

The present manual is intended for a Secure VoIP Service Administrator. It explains how to manage the day-by-day procedure for:

- Create and manage Secure VoIP users and groups
- Perform Automatic Activations through Provisioning
- Manage Secure Conference Rooms
- Create SIP Trunks to connect our PrivateServer appliance to other PBXes
- Handle with the Administrative Roles and Users
- Find and read the logs

Intended audience

This guide is intended for PBX administrators who will provide secure call service by PrivateServer. No specific background is needed

This manual won't explain how to install the server itself. For such knowledge please refer to the **PrivateServer Administration 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.

Secure Call

A **Secure Call** is a voice connection which can't be wiretapped. It runs on the Voice Over IP (VoIP) communication protocol and can be used in two classification 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 non-secure **PBX**. In this case the call is secured in the **PrivateGSM** to **PrivateServer** leg but is not secured on the **PrivateServer** to **PBX** one.

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 secure 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 Rooms

The **conference call** is a call in which more than one person can partecipate. The conference calls are usually defined as "**rooms**" which access can be limted 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.

SIP Accounts

The **SIP Accounts** are the users account on the PrivateServer. Without configuring such accounts none can place or receive **Secure Calls**.

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.

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.

PSUM 1.0 Accounts and Groups