

PrivateGSM - Introduction

PrivateGSM enables you to make phone conversations securely and in complete privacy on your mobile phone.

PrivateGSM exists in two main editions:

- **PrivateGSM Enterprise** can be used on the company network equipped with **Enterprise VoIP Security Suite** and a locally installed **PrivateServer**
- **PrivateGSM Professional** can be used to try out the software in an easy way. It does not require any server configuration. Once installed on the device **PrivateGSM Professional** is able to encrypt all incoming and outgoing calls from/to other **PrivateGSM** users. **In other words the software must be installed on both devices of the caller and the called party.** **PrivateGSM Professional** allows you to invite other users to use the system through the "**Invite others**" feature.

PrivateGSM uses **VoIP** technology (Voice over IP) and requires Internet access.

This guide provides you a complete overview of all the features and scenarios using both **PrivateGSM Enterprise** and **PrivateGSM Professional**.

Requirements

PrivateGSM works on **VoIP** technology (Voice over IP) and thus an internet access is mandatory.

Goals

The present manual will explain to you how to:

- install and update **PrivateGSM**
- configure the client
- perform secure calls
- send and receive secure messages

Intended audience

This guide is intended for advanced users or customer care operators that would work directly on customers' devices. The reader should also read each Quick Start Guide. No particular background is needed.

The present manual won't explain anything about how the service works or what's under the client's hood. For a better understanding of the service you should read [PrivateServer - User Manual](#) as well.

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 participate. 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 participants**. The third user is added to a precedent running conversation by one of the two participants.

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 participant can hold on his/hers peer and perform a new call to the number to which transfer 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 transferred 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.

Secure Message

Secure Message is a text message that can be sent and received only by using PrivateGSM and that shares the same communication infrastructure of Secure Call. The maximum length of each message is 160 character.

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.

[PrivateGSM installation pre-requisites](#)