

Linphone based Video Door Entry Intercom System

Solution description

Office:

Le Trident Bat D - 34, avenue de l'Europe
38100 Grenoble - France
Tel. : +33 (0)9 52 63 65 05

Headquarters:

12, allée des Genêts
38100 Grenoble - France
Tel. : +33 (0)9 52 63 65 05

Company legal information:

SARL au capital de 4000 €
SIRET : 520 318 437 00016
EU VAT Number : FR89 520 318 437

Intro

Digital video door entry intercom systems combined with smartphones can highly leverage on regular SIP VoIP technology to bring a new set of services to end-users: get notified when a visitor presses the ring button, see him with video, interact with him with voice and video - at home as well as anywhere else under wifi or 3G coverage. This document describes how both Linphone and Flexisip can be used together to build a SIP network dedicated to home automation and surveillance.

Index

[Intro](#)

[Index](#)

[In house integration](#)

[Extension to smartphones connected to the public internet](#)

[Smartphone app](#)

[Scenario 1 : Answering a call coming from video entry phone](#)

[Scenario 2 : Checking the video camera of the video entry phone](#)

[Entry video phone](#)

[SIP call flow](#)

[Software development](#)

[Application development](#)

[Hardware integration](#)

[Software packaging](#)

[In-house control panel](#)

[SIP call flow](#)

[Software description](#)

[Administration](#)

[SIP proxy on the public internet](#)

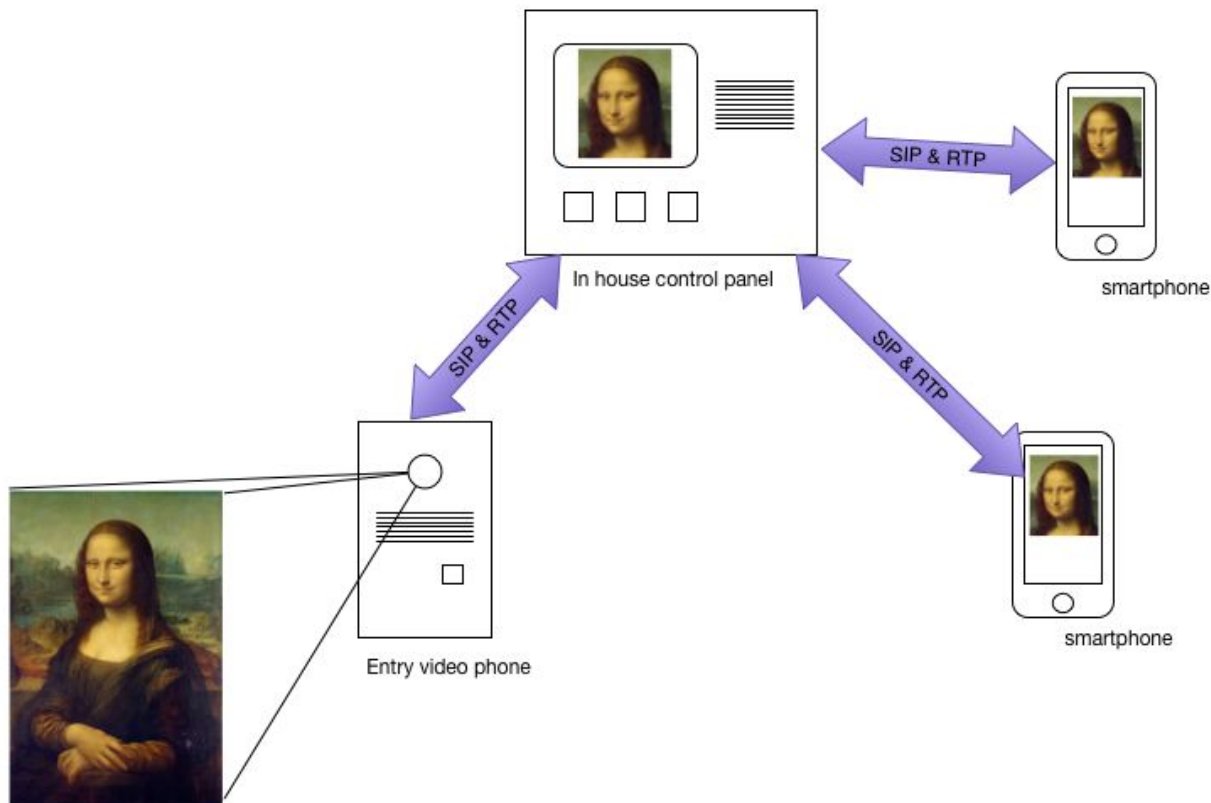
[SIP call flow](#)

[Conclusion](#)

[Contact](#)

In house integration

Digital video door entry intercom systems aims to bring audio and video from an entry phone to both an in house control panel and any smartphones connected either from local network or public internet using regular SIP (rfc3261) & RTP (rfc3550).

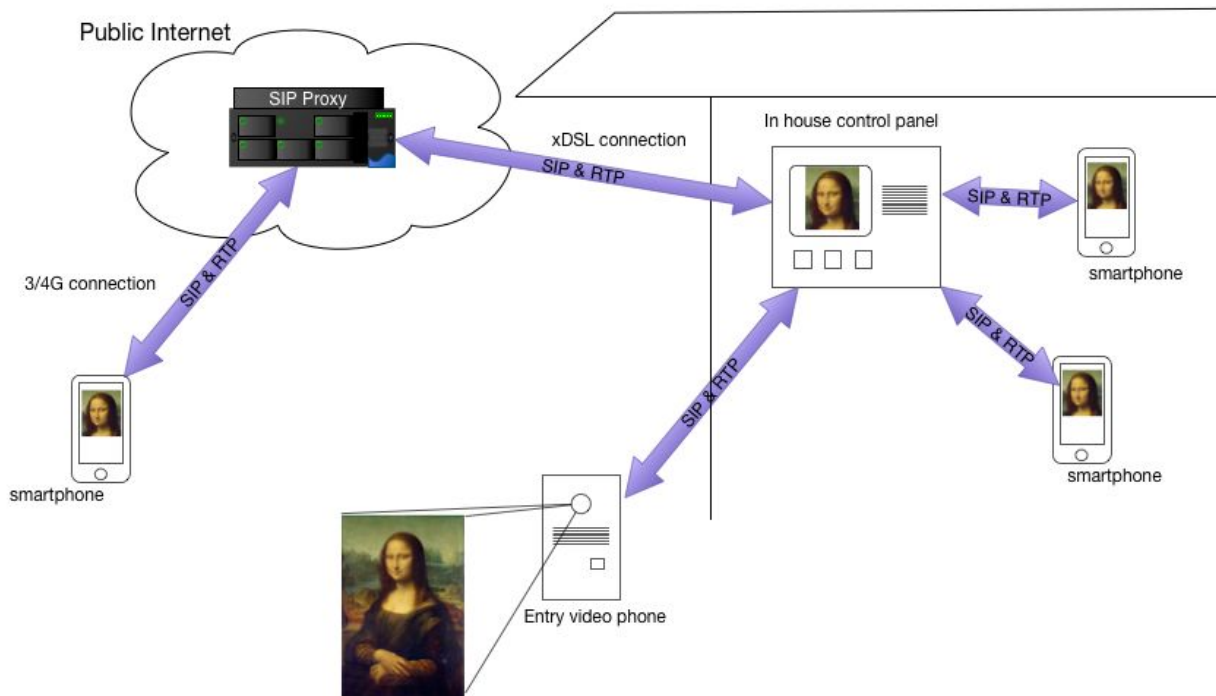


This drawing shows a typical residential deployment where Linphone Voice and Video SDK can be used both for smartphone's apps, embedded in entry video phones and into in-house control panel. This multi-devices deployment requires to use a SIP Proxy server to route SIP video calls from entry video phone to both in-house control panel and smartphone's apps. This call routing functions can be easily be handled by Flexisip, the opensource SIP proxy server written by the Linphone team.

Smartphone's apps also have ability to establish a SIP call to entry door phone in order to monitor video camera at any time.

Extension to smartphones connected to the public internet

A common use case is to extend display of video coming from an entry phone to multiple devices including smartphones connected to public internet.



The above diagram shows a use case where a Flexisip SIP proxy server is deployed in the cloud to serve as audio/video broker between smartphones connected from the public internet and entry video phone connected to the in house Flexisip SIP proxy server embedded in a control panel.

Smartphone app

Scenario 1 : Answering a call coming from video entry phone

When someone rings the doorbell of entry video phone, it initiates an incoming call with a preview video, awaking the app if it was in background mode. The smartphone user can accept the incoming call. (figure 1)

If the user accepts the call, then he can make different actions like opening the door, turning on the mic, switching on the light, hanging up... (buttons are customizable) (figure 2)

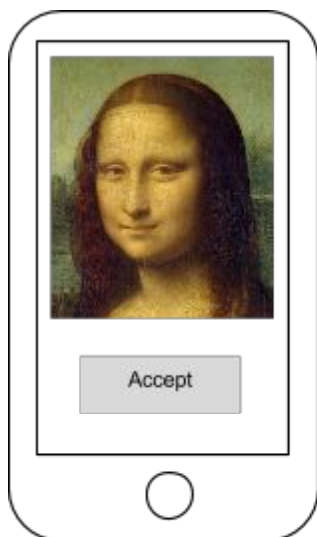


Figure 1

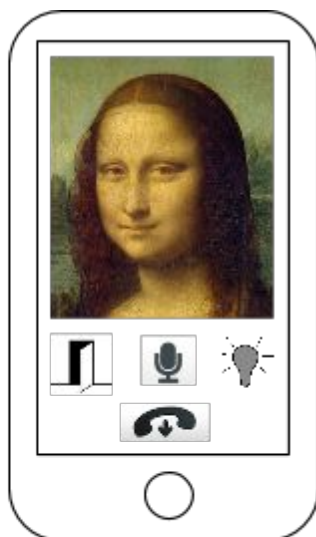
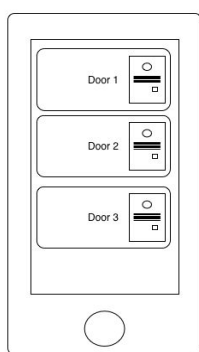


Figure 2

Scenario 2 : Checking the video camera of the video entry phone

At any time a user has the ability to reach entry door phones of his house. When launching the app, a list of available doors is displayed. Pressing on a row establishes a regular SIP call to this entry door phone:

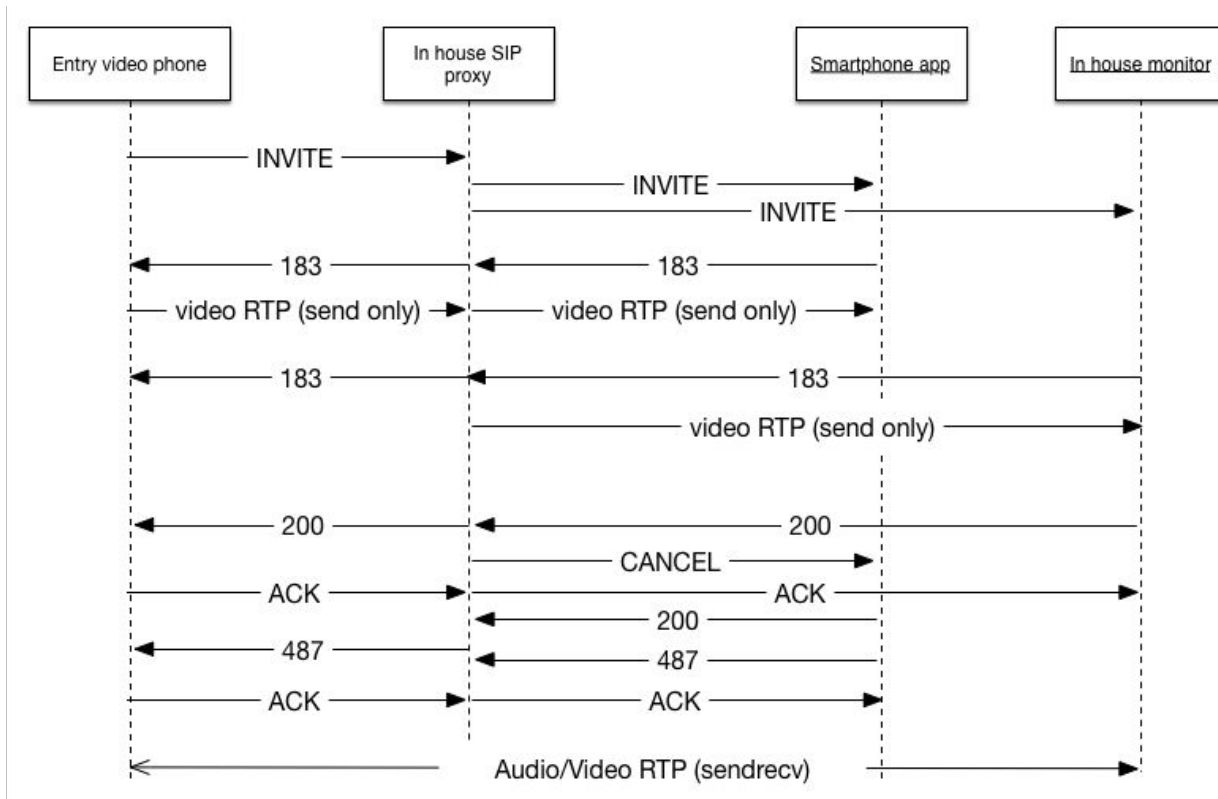


Entry video phone

Entry video phones are typically embedded devices with both a camera and a microphone running Linux. Purpose of this equipment is to initiate a video call when a visitor press the ring button. This video call is started with a one direction early media video and/or audio stream from the entry video phone. Thanks to these early-media streams, it is possible to visualize who is in front of the door during the ringing state. A receiver application either from smartphone app or from in house control panel can decide to accept call to enter in a duplex audio video session or just to unlock the door.

SIP call flow

This use case can be traduced in the following SIP call flow:



- When a visitor arriving at the entry video phone presses the ring button, a SIP INVITE is sent to in house SIP proxy. This SIP proxy, that may eventually be located inside the in-house control panel, then forks this INVITE to all currently registered devices, a smartphone and the in house monitor in this case.
- Both smartphone apps and in house monitor start ringing and answering a 183 SIP response with an SDP indicating a receive-only media direction. These 183 messages are modified by the in-house SIP proxy to insert a media relay into the media path. Purpose of this media relay is to fork media streams coming from entry video phone to all ringing devices.
- Upon reception of first 183, entry phone starts sending video to the SIP proxy which relays media to the smartphone.
- When the SIP proxy receives the second 183 from the monitor, it starts to fork media streams coming from the entry phone to the in-house monitor. Note this second 183 does not have any impact on the entry phone.

- Later, someone in the house may decide to take to call from the in house monitor. This action triggers a 200 ok to be sent to accept the call in full duplex mode.
- Upon receipt of this 200 ok, the in-house SIP proxy cancels all the call branches still in ringing state.

Software development

Application development

Liblinphone VOIP SDK is available in C and Python, which for two languages particularly suitable for embedded systems. For reference documentation, visit [linphone web site](#).

Hardware integration

Liblinphone VOIP SDK leverages on linux for both audio video capture and rendering. On Linux, liblinphone built in interfaces are:

- v4l2 for video capture.
- X11+Xv or OpenGL for video display.
- Alsa, Pulse audio and OSS for audio capture/playback.
- Software echo canceler.

Thanks to a plugin API, Liblinphone VOIP SDK media processing engine Mediastreamer2 can be easily extended to interface with hardware based video encoder or echo canceler. For reference documentation, go to [Mediastreamer2 API documentation](#).

Software packaging

A BitBake based distribution is planned to specially address embedded system like Yocto Linux. Alternatively, regular rpms can also be used.

In-house control panel

In-house control panel is very often the brain of an entry door system. This is perfect place for the in-house SIP Proxy to seat.

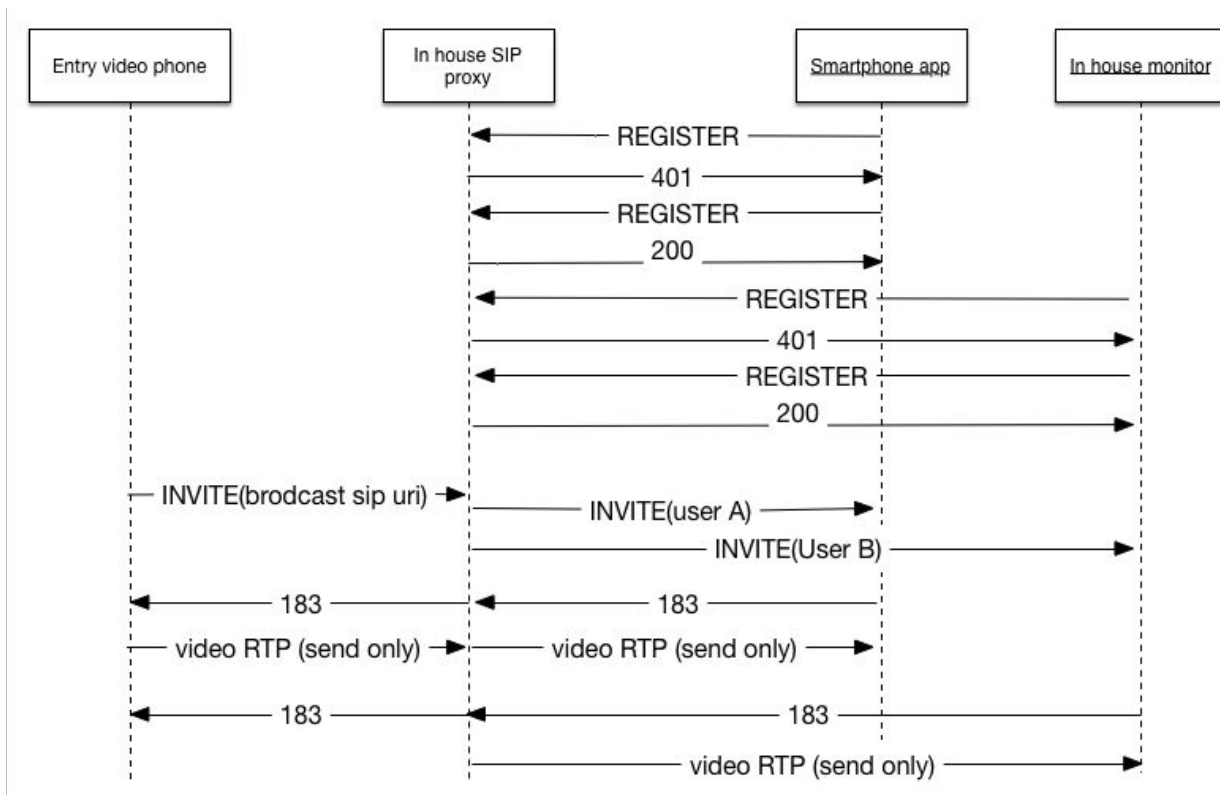
Purpose of this SIP proxy is to route incoming calls arriving from entry phones to all appliances in the house, namely smartphone's app and in-house monitor. In-house SIP proxy main functions are the following:

- Registrar as defined by rfc3261 to be able to reach smartphone's apps.
- User authentications to make sure only authorized smartphone's app can received incoming call from entry phones.
- Call forking to distribute incoming calls to all connected devices.
- Media relay forking to replicate media stream from entry phones to all ringing devices.
- Portability on embedded devices including ARM based devices.
- Ability to associate a single sip address to all users registered to a sip domain (for group calling).

Additionally, control panels with video display can also be based on liblinphone VOIP sdk. This case is similar to [Entry video phone](#).

SIP call flow

This use case can be traduced in the following SIP call flow:



- Once connected to in house wifi network, a SIP REGISTER for a given domain is sent from smartphone app to in house SIP proxy. This REGISTER is challenged
- Smartphone app initiates a subsequent REGISTER with authorization info which are checked by in-house SIP proxy.
- Same process is repeated every time a new appliance connects to homenetwork.
- When someone presses the doorbell, INVITE message with a broadcast sip uri is received by the in house SIP proxy. This invite is forked to all users registered for this SIP domain.

Flexisip, the opensource SIP proxy server written by the Linphone team was developed with these use cases in mind.

Software description

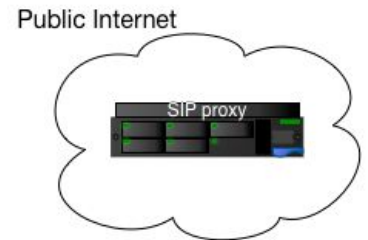
Flexisip can be compiled and deployed on any Linux based distribution, both ARM and x86. Currently both deb and rpm packages are available. A BitBake based distribution is planned to specially address embedded system like Yocto Linux.

Administration

Flexisip is packaged as a Linux service with start/stop function. Debug traces are logged into syslog. Configuration take place in a file in /etc/flexisip. For administration details, refer to the [Flexisip documentation](#).

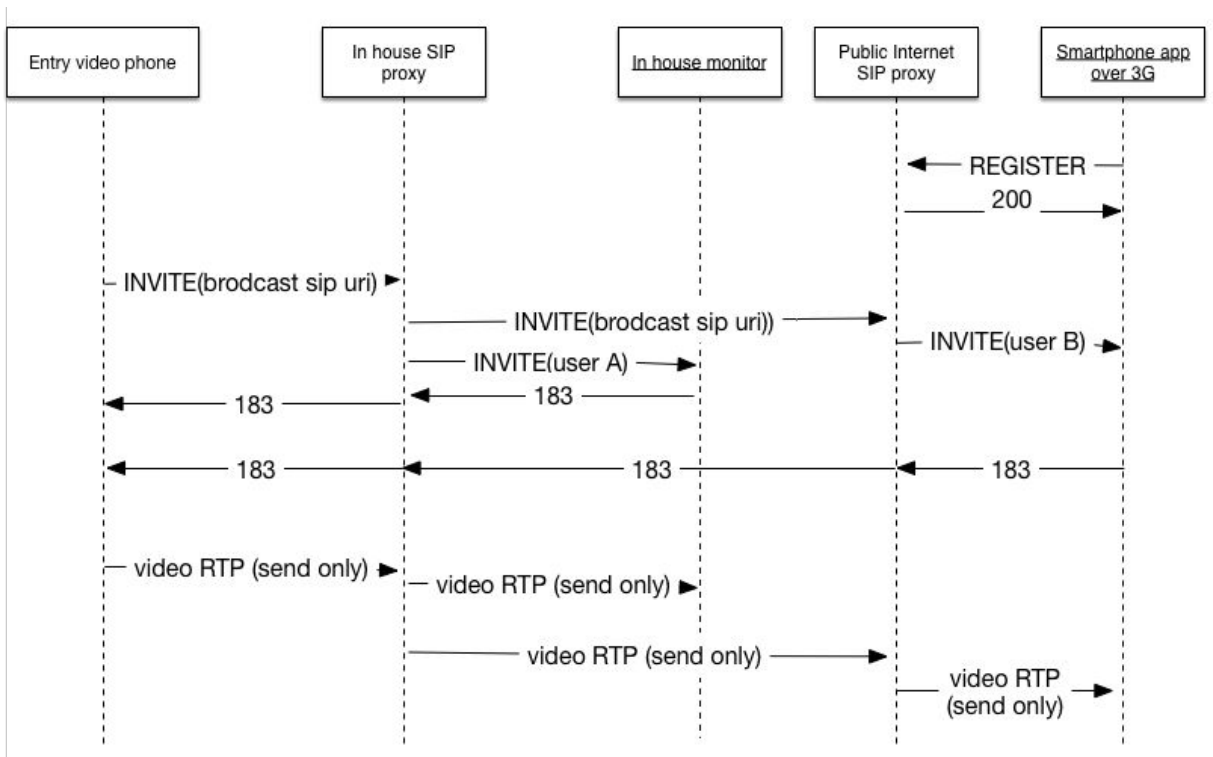
SIP proxy on the public internet

An IP based door entry intercom systems shall always be able to reach smartphone's apps, even if not connected to in house network. This function is implemented by a SIP proxy server in the cloud that can be reached from a mobile device connected to public internet. This SIP proxy server shares its specification with the in house proxy but with addition of some specific features which are outlined below :



- Secure connection based on SIP-TLS and SRTP.
- Push notification integration to maximize smartphone reachability.
- Scalability/high availability to be a central point for all subscribers.
- Multi SIP domain management grouping users by houses.
- Manages in-house SIP proxy connections for interconnection home network.

SIP call flow



- Smartphones over 3G or public WIFI register to public internet SIP proxy using a secure connection, under the SIP domain of their home network.
- When a visitor arrives at entry video phone and ring button is pressed, a SIP INVITE is sent to in house SIP proxy. This SIP INVITE is forwarded to both local users registered in the house and to the public internet SIP proxy.
- Public internet SIP proxy transfers received INVITE to all registered users for a given home SIP domain.
- Video RTP sent from entry video phone is forked by the in house SIP proxy to both users having answered 183 responses in house, and to public internet SIP proxy if a 183 was received from outside.
- Public internet SIP proxy forks to all users having answered 183.

Conclusion

This document demonstrated relevance of adapting a voice and video over IP architecture to digital intercom systems. Linphone and Flexisip is the perfect couple to address communications needs for both in and outside of house. Belledonne Communications is able to adapt this reference architecture to customer's specific needs.

Contact

Worldwide sales

sales@belledonne-communications.com

Office

Belledonne Communications
Le Trident Bat D - 34 avenue de l'Europe
38100 Grenoble - FRANCE

Tel +33 (0)9 52 63 65 05

Fax +33 (0)9 57 63 65 05

info@belledonne-communications.com