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communications.

# Linhome-based Video Door Entry Intercom System Solution description

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# Intro

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Digital video door entry intercom systems combined with mobile devices 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 Wi-Fi or with mobile network coverage. This document describes how both Linhome, liblinphone sdk and Flexisip can be used together to build a SIP network dedicated to home automation and surveillance.

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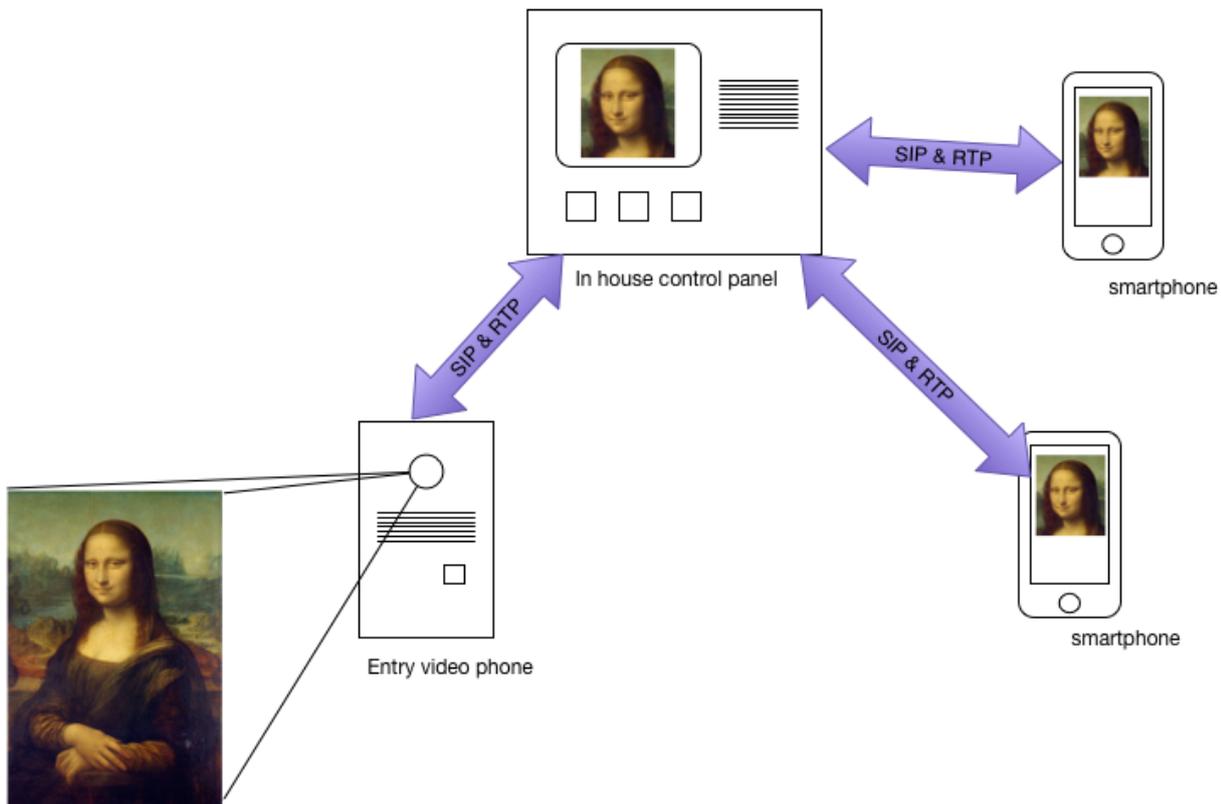
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# Solution architecture

## In house integration

Digital video door entry intercom systems aim to bring audio and video from an entry phone to both an in house control panel and any mobile device that is connected either via a local network or the public internet, using regular SIP (rfc3261) & RTP (rfc3550).

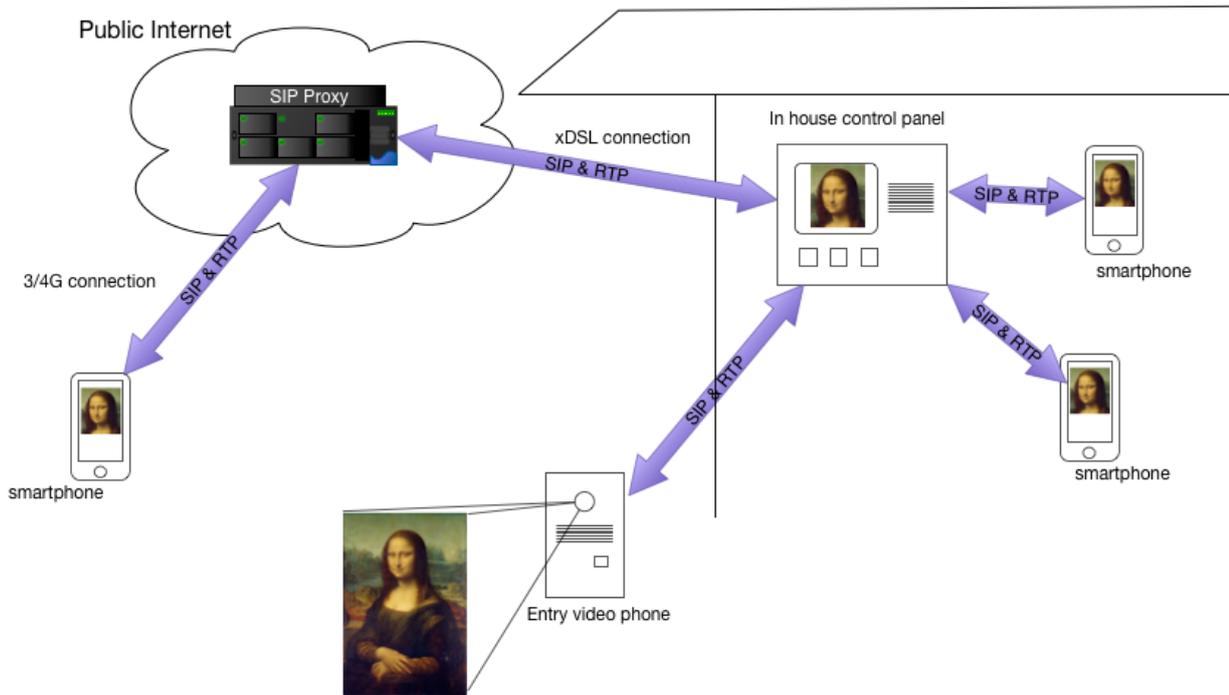


This drawing shows a typical residential deployment where liblinphone Voice and Video SDK can be used both for mobile apps, embedded in entry video phones and into in-house control panel. This multi-devices deployment requires to use a SIP Proxy server to route SIP video calls from the entry video phone to both the in-house control panel and mobile apps. This call routing function can easily be handled by Flexisip, the open source SIP proxy server written by the Belledonne Communications, the company developing both Linphone and Linhome apps.

Mobile apps also have the ability to establish a SIP call to an 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 the display of the video coming from an entry phone to multiple devices, including smartphones connected to the 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 an entry video phone connected to the in-house Flexisip SIP proxy server that is embedded in a control panel.

# Solution components

## Mobile app

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Mobile apps have been developed to meet the needs for end-users to both control and monitor audio and video doorbells from their smartphone or tablet/iPad.

Main features are:

- The ability to be notified of someone ringing at the door, with video preview if available. It includes the ability to start an audio call to communicate with the visitor.
- Outgoing calls to preconfigured sip-based doorbells or video cameras.
- Action buttons to switch on the light, or any possible doorbell's action to be remotely controlled.
- Call history with recording of audio/video streams coming from the doorbell.
- Easy QR-code based remote configuration.

### Typical use cases:

#### Scenario 1: Answering a call coming from a video door phone

When someone is ringing the doorbell of an entry video phone, it initiates an incoming SIP call with video preview, which awakes the app if it is running in the background, using Apple's and Google's VoIP push notifications (figure 1&2).

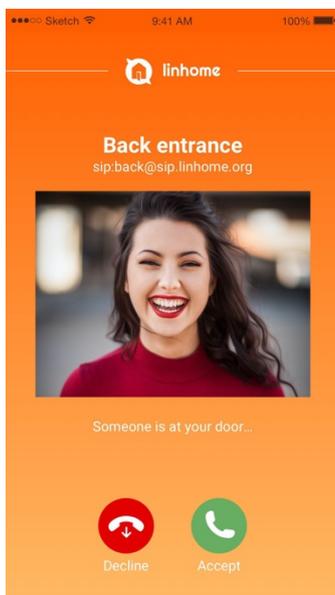


Figure 1 - Linhome incoming call view

(the Linhome app was running in the foreground)

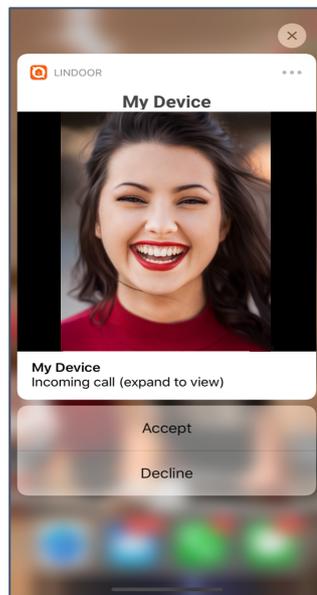


Figure 2 - Linhome push notification view (iOS)

(the Linhome app was running in the background)

The smartphone's user can accept the incoming call (figure 3&4). If the user accepts the call, then he can make different actions like opening the door, turning on the microphone, switching on the light, hanging up, etc. (action buttons are customizable)

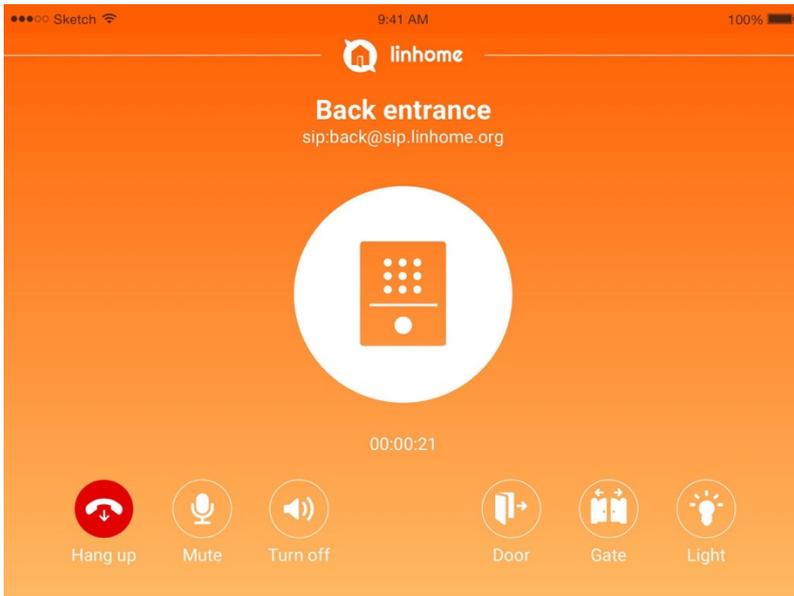


Figure 3 Linhome 2-ways Audio communication

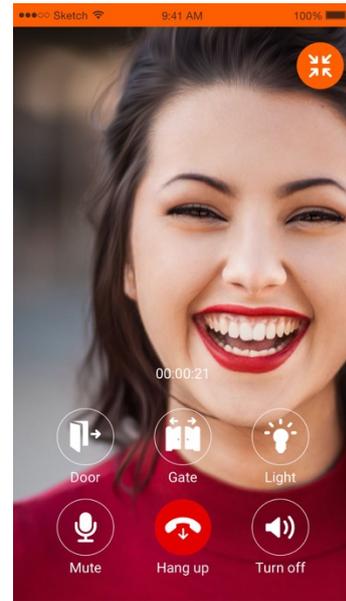


Figure 4 – Linhome Video communication

## Scenario 2 : Checking the video camera of the doorbell

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

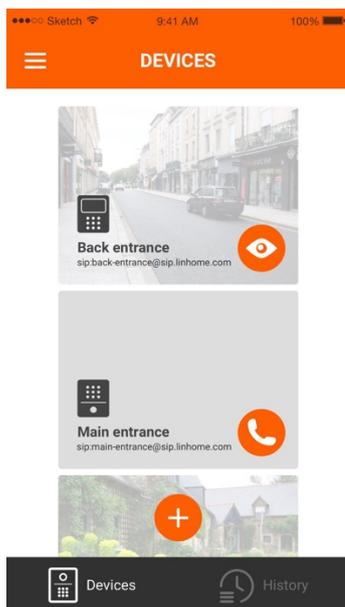


Figure 5 – Linhome list of home's devices

## Application details

Belledonne communications develops and distributes Linhome, a full-featured highly customizable mobile application designed for intercom use cases. Written in Swift/Kotlin, it's available on both IOS and Android.

Linhome is always reachable thanks to its implementation of Push Notification for SIP (rfc8599).

Linhome offers a built-in integration with a Flexisip sip proxy backend. It can also be used with any third party SIP server. For reference, visit our Linhome's web site at [www.linhome.org](http://www.linhome.org).

Linhome leverages the liblinphone's VOIP SDK. For reference, got to: [www.linphone.org/technical-corner/liblinphone](http://www.linphone.org/technical-corner/liblinphone).

## Compatibility with intercom hardware products

Linhome is compatible with any SIP-based entry door phone. Main supported standards are:

- SIP, for session establishment
- Early media, for video preview
- H264/G711, as primary audio/video codecs
- RTP secured/unsecured with or without AVPF extension

## Entry video phone

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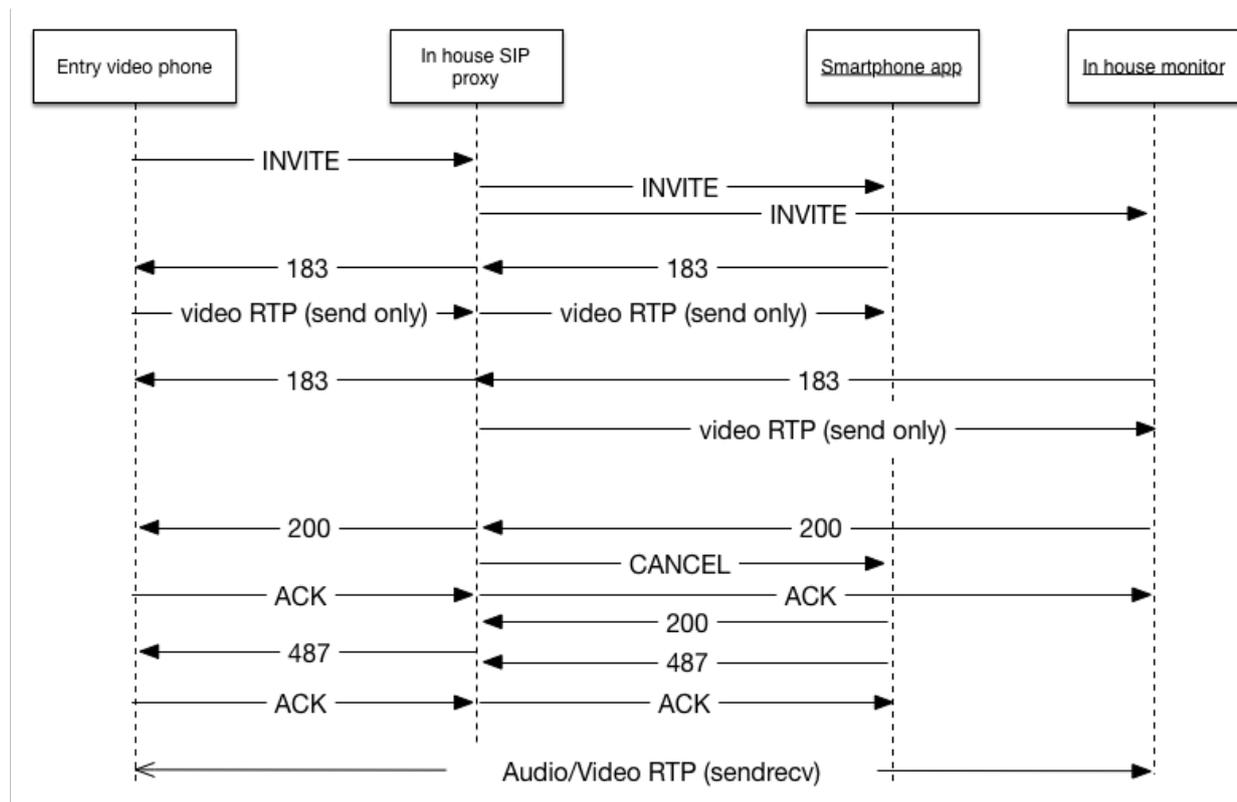
Entry video phones are typically embedded devices with both a camera and a microphone, running Linux. Purpose of this equipment is to initiate a video call when a visitor presses the ring button. This video call is started with a one-way early-media video stream and/or a two-ways audio stream sent by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

### Typical use cases:

#### Scenario 1: Answering a call coming from a video entry phone

##### SIP call flow

This use case corresponds to the following SIP call flow:



- When a visitor presses the entry video phone’s ring button, a SIP INVITE is sent to the 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, in this case a mobile device (a tablet or a smartphone) and the in-house monitor.
- Both the mobile app and the 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 the entry video phone to all ringing devices.
- Upon reception of first 183, the entry phone starts sending video to the SIP proxy that relays media to the mobile device.
- When the SIP proxy receives the second 183 from the monitor, it starts forking 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 the call from the in-house monitor. This action triggers a 200 ok, which is 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 that are still in ringing state.

### Application development

Liblinphone VOIP SDK is available in C, C++ and Python, which are languages particularly suitable for embedded systems. For reference documentation, visit [linphone web site](http://www.linphone.org).

### Hardware integration

Liblinphone VOIP SDK leverages linux for both audio and video capture and rendering. On Linux, liblinphone’s 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's media processing engine, called Mediastreamer2, can easily be 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 specifically address embedded systems, such as Yocto Linux. Alternatively, regular rpms can also be used.

## In-house control panel

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In-house control panels are very often the brain of an entry door system. They are the perfect place for the in-house SIP Proxy to seat.

The purpose of this SIP proxy is to route incoming calls arriving from entry phones to all appliances in the house, namely mobile apps and in-house monitors. The main functions of the in-house SIP are the following:

- Registrar, as defined by rfc3261, to be able to reach mobile apps.
- User authentication, to make sure that only authorized mobile apps can receive incoming calls from the entry phones.
- Call forking, to distribute incoming calls to all connected devices.
- Media relay forking, to replicate media streams sent by the entry phones to all ringing devices.
- Portability on embedded devices, including ARM based devices.
- Ability to associate a single sip address to all users who are registered to the same sip domain (for group calling).

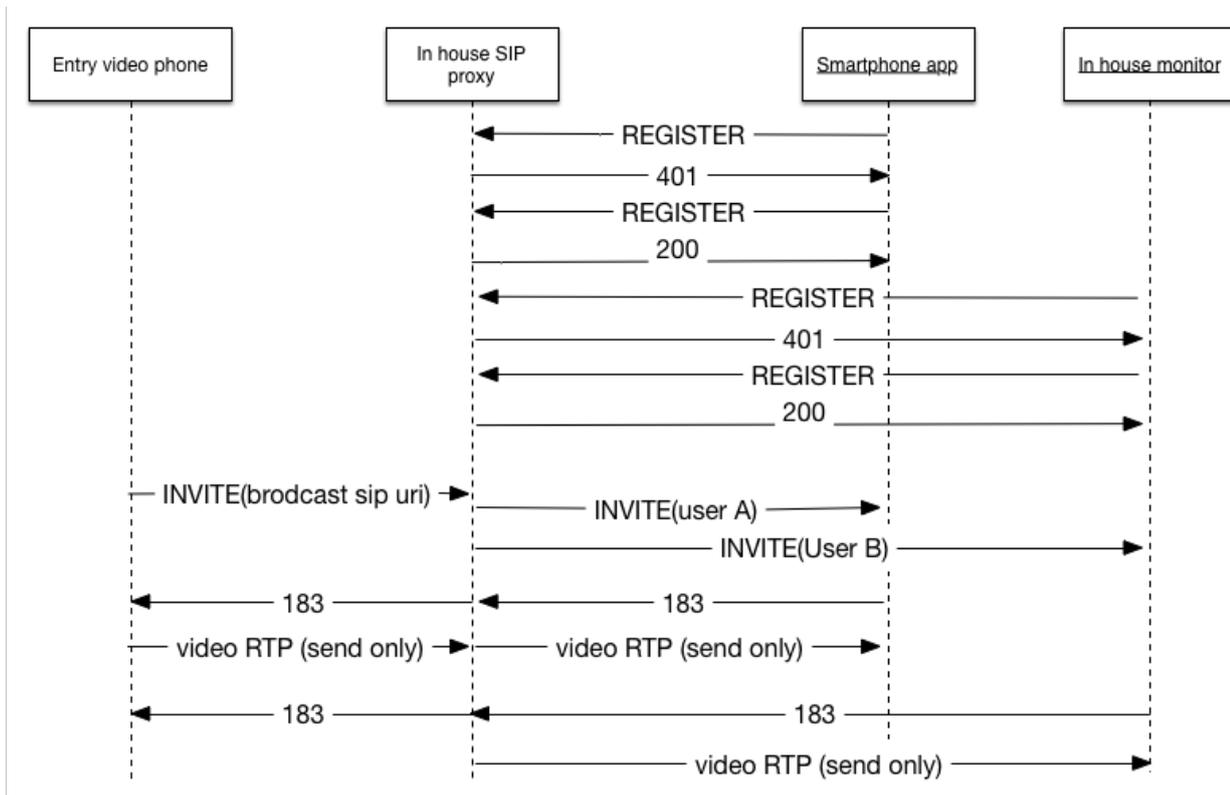
Additionally, control panels with video display can also be based on liblinphone VOIP sdk. This case is similar to [Entry video phone](#).

### Typical use cases:

**Scenario : a call coming from a video entry phone is displayed on both an in-house monitor and a mobile app.**

SIP call flow

This use case corresponds to the following SIP call flow:



- Once connected to the local Wi-Fi network, a SIP REGISTER for a given domain is sent from the mobile app to the in-house SIP proxy. This REGISTER is challenged.
- The mobile app initiates a subsequent REGISTER with authorization info, which are checked by the in-house SIP proxy.
- The same process is repeated every time a new appliance connects to the home network.
- When someone presses the doorbell, an INVITE message with a broadcast sip uri is received by the in-house SIP proxy. This INVITE is forked to all users registered to this SIP domain.

Flexisip, the open source SIP proxy server that is written by the Linhome team, was developed with these use cases in mind.

## Software description

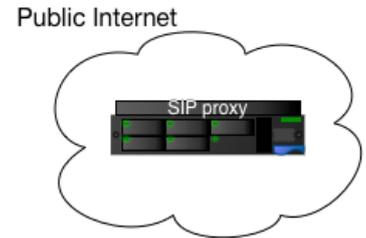
Flexisip can be compiled and deployed on any Linux based distribution, both ARM and x86. Currently both deb and rpm packages are available. A BitBake based distribution is planned to specifically address embedded system, such as Yocto Linux.

## Administration

Flexisip is packaged as a Linux service with start/stop functions. 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 system shall always be able to reach mobile apps, even if the mobile devices on which the app is running are not connected to the in-house local network. This function is implemented by a SIP proxy server deployed in the cloud (the “cloud SIP proxy”), that can be reached from a mobile device connected to the public internet. This cloud SIP proxy server shares its specification with the in-house proxy, but with addition of specific features, as listed below:

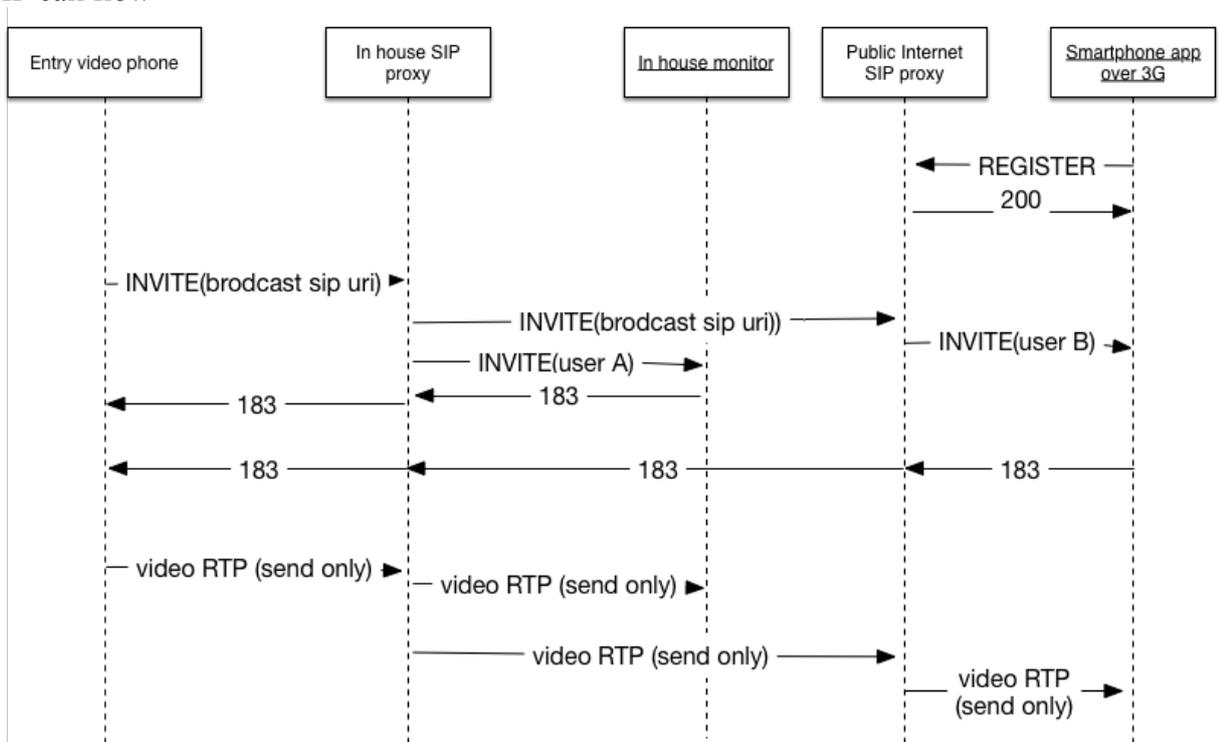


- Secure connection based on SIP-TLS and SRTP.
- Push notification integration, to maximize smartphone reachability.
- Scalability/high availability, to be a central point for all subscribers.
- Multi SIP domain management grouping users by houses.
- Management of in-house SIP proxy connections, for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

## Typical use cases:

**Scenario :** A all coming from a video entry phone is displayed on both an entry video phone and a mobile app via the public internet.

### SIP call flow



- A mobile device that is connected to the public internet – either over 3G/4G or public WI-FI – registers to the cloud SIP proxy using a secure connection, under the SIP domain of their home network.
- When a visitor rings the bell button, a SIP INVITE is sent to the in-house SIP proxy. This SIP INVITE is forwarded to both local users registered in the house and to the cloud SIP proxy.
- The cloud SIP proxy transfers the received INVITE to all registered users for a given home’s SIP domain.

- The video RTP stream sent from the entry video phone is forked by the in-house SIP proxy to both users having answered 183 responses in house, and to the cloud SIP proxy if a 183 was received from outside.
- The cloud SIP proxy forks the video RTP stream to all users having answered 183.

## Conclusion

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This document demonstrated relevance of adapting a voice and video over IP architecture to digital intercom systems. Linhome, liblinphone SDK 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 specifics needs.

## Contact

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