
# Real-time audio/video conferences in Linphone thanks to a modern SFU server

presented by Jehan Monnier

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linphone
open source VOIP project

# Presentation

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**Function:** Cofounder of Belledonne Communications
Involved in Linphone's development since 2010

Belledonne Communications is the company behind the **Linphone** project

# Agenda

# I. LINPHONE INTRODUCTION

# Quick intro

## Linphone

- Is around since 2001
- Is available on Linux, android, iOS, Windows, Mac
- Uses SIP standards for audio, video and instant messaging
- Secure group messaging using a Signal protocol derivative

## Linphone's team also provides

- Flexisip, an open source SIP Proxy
- A free SIP service sip.linphone.org

# II. VIDEO CONFERENCE WITH SIP

# Video conference according to IETF and GSMA/RCS specifications

## Session establishment

- SIP methods: INVITE, REFER, BYE between clients and a conference server
- SIP Call Control - Conferencing for User Agents – RFC4579
- Conference Establishment Using Request-Contained Lists in SIP – RFC5366
- Obtaining and Using Globally Routable User Agent URIs (GRUUs) in SIP – RFC5627 : working with multiple devices

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## Event notification to clients

*(someone joined or left the group etc.)*

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- SIP Event Package for Conference State - RFC4575

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  - Bundle mode RFC8108

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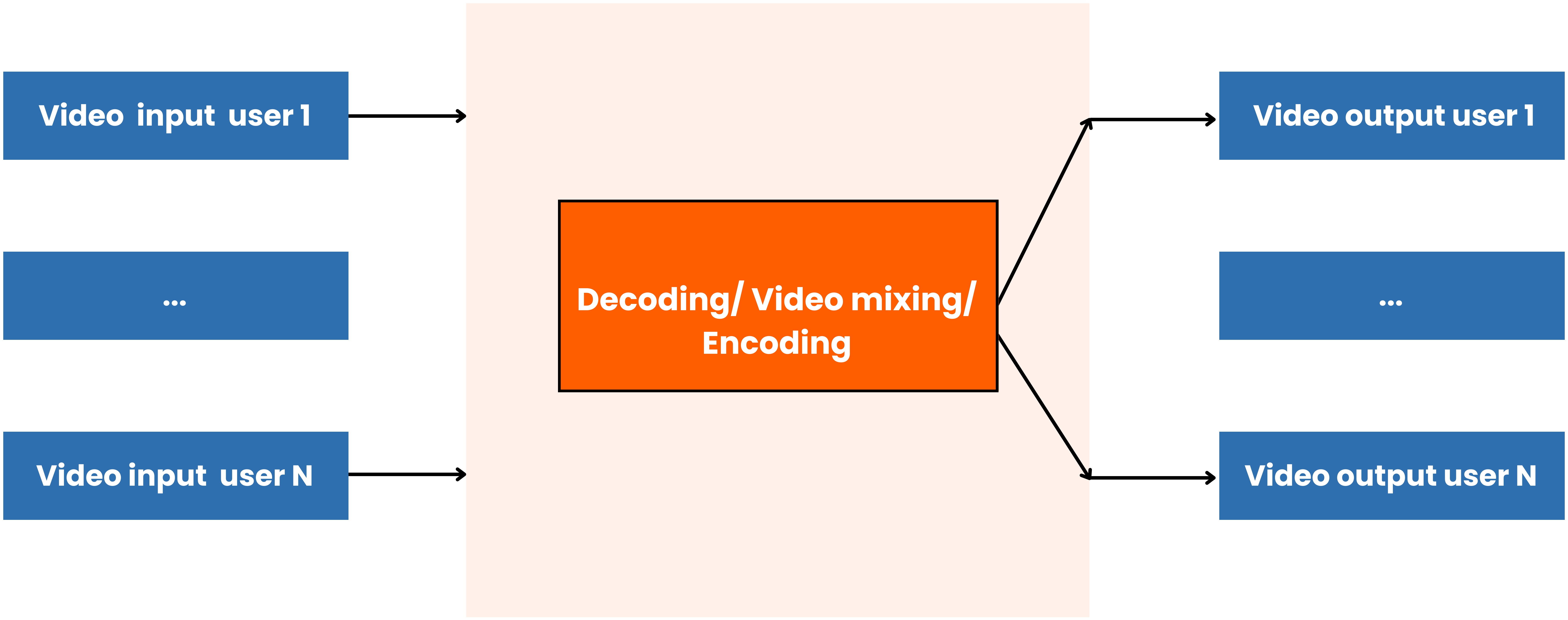
## Security

- SIPS
- SDES, ZRTP, SRTP-DTLS
- AES 256

# Legacy video Conferencing serveur



# RTP view



```
+-A----------+          +-MIXER-----------------------+
| +-RTP1-----|          |-RTP1------+        +-----+ |
| | +-Audio-|          |-Audio---+ | +---+  |     | |
| | |    AA1|--------->|---------+-+-|DEC|->|     | |
| | |       |<---------|MA1 <----+ | +---+  |     | |
| | |       |          |(BA1+CA1)|\| +---+  |     | |
| | +-------|          |---------+ +-|ENC|<-| B+C | |
| +---------|          |-----------+ +---+  |     | |
+------------+          |                    |     | |
                        |                    |  M  | |
+-B----------+          |                    |  E  | |
| +-RTP2-----|          |-RTP2------+        |  D  | |
| | +-Audio-|          |-Audio---+ | +---+  |  I  | |
| | |    BA1|--------->|---------+-+-|DEC|->|  A  | |
| | |       |<---------|MA2 <----+ | +---+  |     | |
| | +-------|          |(AA1+CA1)|\| +---+  |     | |
| +---------|          |---------+ +-|ENC|<-| A+C | |
+------------+          |-----------+ +---+  |     | |
                        |                    |  M  | |
+-C----------+          |                    |  I  | |
| +-RTP3-----|          |-RTP3------+        |  X  | |
| | +-Audio-|          |-Audio---+ | +---+  |  E  | |
| | |    CA1|--------->|---------+-+-|DEC|->|  R  | |
| | |       |<---------|MA3 <----+ | +---+  |     | |
| | +-------|          |(AA1+BA1)|\| +---+  |     | |
| +---------|          |---------+ +-|ENC|<-| A+B | |
+------------+          |-----------+ +---+  +-----+ |
                        +-----------------------------+
```

Figure 15: Session and SSRC Details for Media Mixer

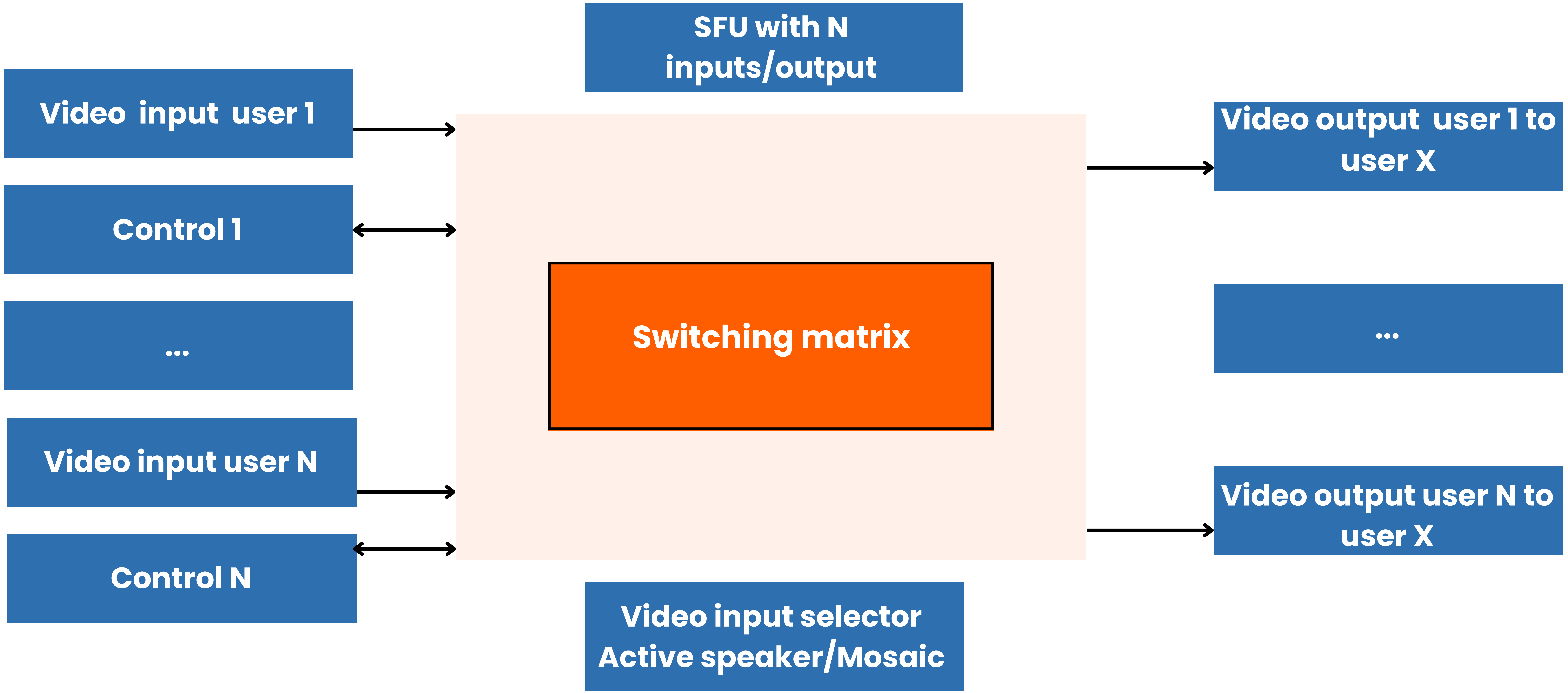
# Pros and Cons

- **Pros**
  - No changes required client side compared to reglar call

- **Cons**
  - Video layout defined server side
  - Huge cpu resource consumption server side
  - No possible e2e encryption

# III. SELECTIVE FORWARDING UNIT

# Selective forwarding Unit



# RTP view

Westerlund & Wenger Informational [Page 29]

RFC 7667 RTP Topologies November 2015

```
+-A---------+             +-Middlebox-----------------+
| +-RTP1----|             |-RTP1------+       +-----+ |
| | +-Video-|             |-Video---+ |       |     | |
| | |    AV1|------------>|---------+-+------>|     | |
| | |       |<------------|BV1 <----+-+-------|  S  | |
| | |       |<------------|CV1 <----+-+-------|  W  | |
| | |       |<------------|DV1 <----+-+-------|  I  | |
| | |       |<------------|EV1 <----+-+-------|  T  | |
| | |       |<------------|FV1 <----+-+-------|  C  | |
| | +-------|             |---------+ |       |  H  | |
| +---------|             |-----------+       |     | |
+-----------+             |                   |  M  | |
                          |                   |  A  | |
+-B---------+             |                   |  T  | |
| +-RTP2----|             |-RTP2------+       |  R  | |
| | +-Video-|             |-Video---+ |       |  I  | |
| | |    BV1|------------>|---------+-+------>|  X  | |
| | |       |<------------|AV1 <----+-+-------|     | |
| | |       |<------------|CV1 <----+-+-------|     | |
| | |       | :    :    : |: :  : : : : :  : :|     | |
| | |       |<------------|FV1 <----+-+-------|     | |
| | +-------|             |---------+ |       |     | |
| +---------|             |-----------+       |     | |
+-----------+             |                   |     | |
                          :                   :     : :
                          :                   :     : :
+-F---------+             |                   |     | |
| +-RTP6----|             |-RTP6------+       |     | |
| | +-Video-|             |-Video---+ |       |     | |
| | |    FV1|------------>|---------+-+------>|     | |
| | |       |<------------|AV1 <----+-+-------|     | |
| | |       | :    :    : |: :  : : : : :  : :|     | |
| | |       |<------------|EV1 <----+-+-------|     | |
| | +-------|             |---------+ |       |     | |
| +---------|             |-----------+       +-----+ |
+-----------+             +---------------------------+
```

Figure 17: Selective Forwarding Middlebox

# Pros and Cons

- **Pros**
    - Video layout defined client side.
    - Low cpu footprint server side.
    - Allow e2e encryption
- **Cons**
    - Requires multi stream management client side

# IV. SIP USER AGENT CHANGES

# SIP User Agent changes

- Multi streams requirements
  - Needs for encoding multiple video stream (Main + thumbnails) or to use multi layered video compressor like H264 AVC/VP9
  - Need for decoding muliple video stream, one per conference participants

linphone

# Sample SDP server side

- v=0
- o=conference-focus 887 2965 IN IP4 5.135.31.161
- a=group:BUNDLE as vs vsBa9vMej1r vs39~J4bdS7
- m=audio 35906 RTP/AVP 96 97 101
- a=rtpmap:96 opus/48000/2
- a=rtcp-mux
- a=mid:as
- a=extmap:1 urn:ietf:params:rtp-hdrext:sdes:mid
- a=extmap:3 urn:ietf:params:rtp-hdrext:csrc-audio-level
- a=extmap:2 urn:ietf:params:rtp-hdrext:ssrc-audio-level vad=off
- a=candidate:1 1 UDP 2130706303 5.135.31.161 35906 typ host
- m=video 0 RTP/AVP 96
- a=rtpmap:96 VP8/90000
- a=recvonly
- a=mid:vs
- a=bundle-only
- a=rtcp-fb:* ccm tmmbr
- a=label:Ba9vMej1r
- a=content:main
- m=video 0 RTP/AVP 96
- a=rtpmap:96 VP8/90000
- a=recvonly
- a=label:Ba9vMej1r
- a=content:thumbnail
- m=video 0 RTP/AVP 96
- a=rtpmap:96 VP8/90000
- a=sendonly
- a=label:39~J4bdS7

# How to test

- Testing server compatible with **RFC4579** is available at:

  **sip:videoconference-factory@sip.linphone.org**

- Linphone client above **5.X** can be used

# V. CONCLUSION

# Possible Evolutions

- XCON for advanced conferencing management
- E2E encryption thanks to SFU
- Web client compatibility with Webrtc

# Linphone Conferencing software


- **3 display modes**
- Meeting scheduling with invitation sending
- Modern open-source video **conferencing solution** based on SIP
- Easy scalability with server-side stream selection (**SFU**)

# Useful links

- **Linphone website**: https://www.linphone.org
- **Mediastreamer2**: https://www.linphone.org/technical-corner/mediastreamer2-ortp
- **Conference-server**: https://wiki.linphone.org/xwiki/wiki/public/view/Flexisip/A.%20Configuration%20Reference%20Guide/master/conference-server
- **Linphone Features**: https://www.linphone.org/features
- **Linphone on gitlab**: gitlab.linphone.org


# Thank you!

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# Do you have any questions ?



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