SylkSuite

@agprojects

Naples 2019-09-24


SDP
We are all sinners

THE START OF THE AFFAIR

affair occurred on a day in 458 BC

How did all start?


m=audio 50006 RTP/AVP 0

```
a=rtpmap:13 CN/8000
a=rtpmap:110 telephone-event/48000
a=rtpmap:113 telephone-event/16000
a=rtpmap:126 telephone-event/8000
a=fmtp:111 minptime=10;useinbandfec=1
a=rtcp:9 IN IP4 0.0.0.0
a=rtcp-fb:111 transport-cc
a=extmap:1 urn:ietf:params:rtp-hdrext:ssrc-audio-level
a=setup:actpass
a=mid:audio
a=sendrecv
a=ice-ufrag:3WW+
a=ice-pwd:C8ExxCTWFxc+2V+Pj/ky8UYZ
a=fingerprint:sha-256 7D:CF:89:09:47:32:9C:DD:68:39:37:84:85:60:2B:53:A8:08:4F:6E:0D:
37:90:53:BE:4F:D1:46:41:06:B3:09
a=ice-options:trickle
a=ssrc:2817869627 cname:13+3yqY7vBEe68ie
a=ssrc:2817869627 msid:16582206-db7a-47ed-98d8-c097a32bafb7
67b6939f-1185-4798-89ee-ee334211cfc1
a=ssrc:2817869627 mslabel:16582206-db7a-47ed-98d8-c097a32bafb7
a=ssrc:2817869627 label:67b6939f-1185-4798-89ee-ee334211cfc1
a=rtcp-mux
m=video 9 UDP/TLS/RTP/SAVPF 96 97 98 99 100 101 127 125 104
c=IN IP4 0.0.0.0
a=rtpmap:96 H264/90000
a=rtpmap:97 rtx/90000
a=rtpmap:98 H264/90000
a=rtpmap:99 rtx/90000
a=rtpmap:100 VP8/90000
a=rtpmap:101 rtx/90000
a=rtpmap:127 red/90000
a=rtpmap:125 rtx/90000
```

Hello, my name is Adrian Georgescu!

and I am a SDPholic

Largest SDP carrier at the time (1996)

SIP v1

Around the world in 10 IETFs

Let’s fix it!


Largest SDP carrier in the world (2002)
SIP v2

SIP Express Router

And the Romanian connection

SIP Express Router

Prof. Dr. Ing. Radu Popescu-Zeletin (born in Romania)

Society for Mathematics and Information Technology Gesellschaft für Mathematik und Datenverarbeitung (GMD)

GMD later integrated into Fraunhofer Society as the Institute for Open Communication Systems

Renamed to **FOKUS** in 2000

FOKUS received two European grants: 6 Net and Evolute

Romanian students from Bucharest Institute of Polytechnics were hired

SIP Beyond VoIP

@agprojects

motto

SIP Express Router

MediaProxy

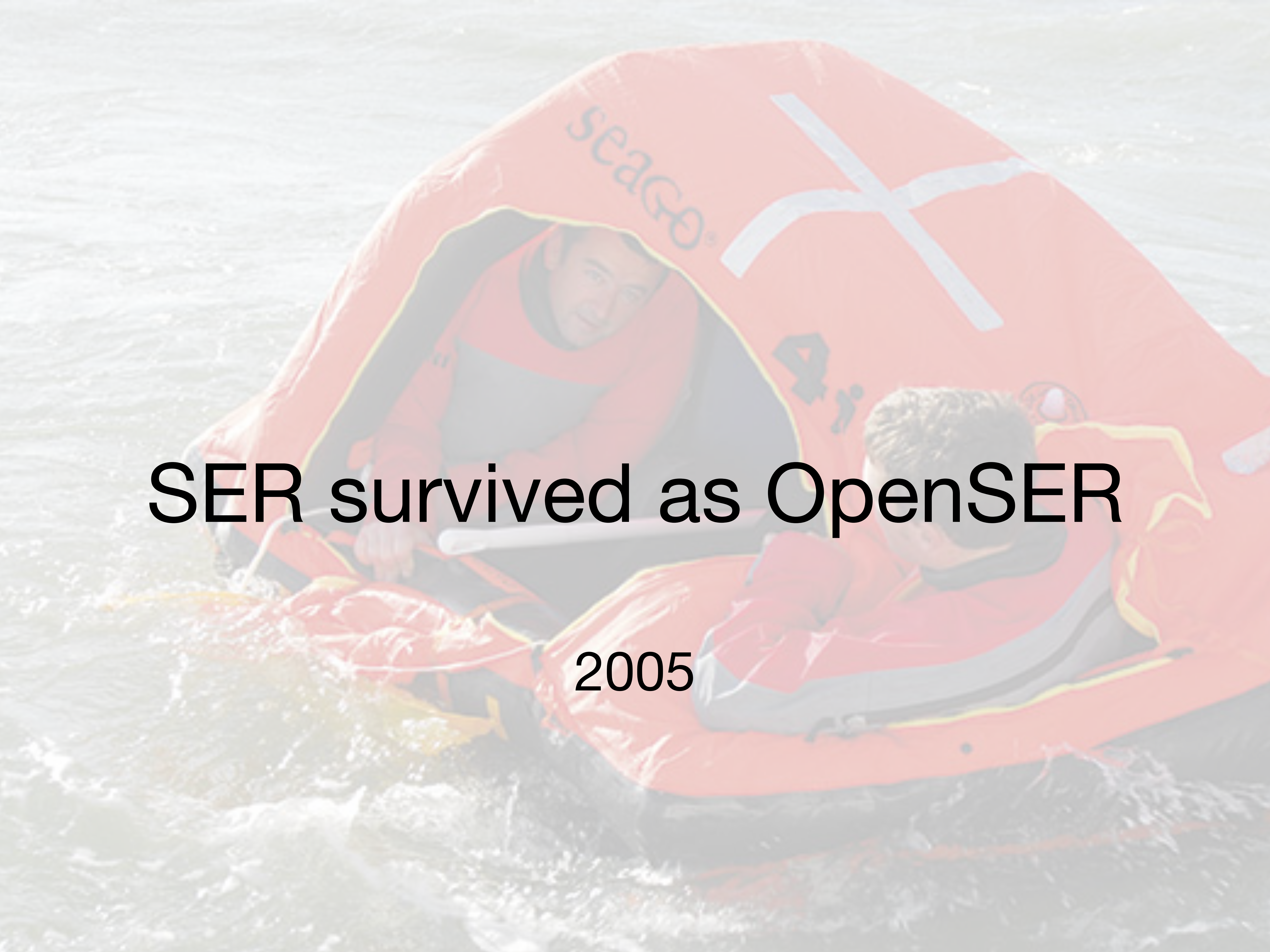
CDRTool

CDRTool is written in PHP, those pesky Haskell programmers still have a choice

Fraunhofer

Fraunhofer-Gesellschaft

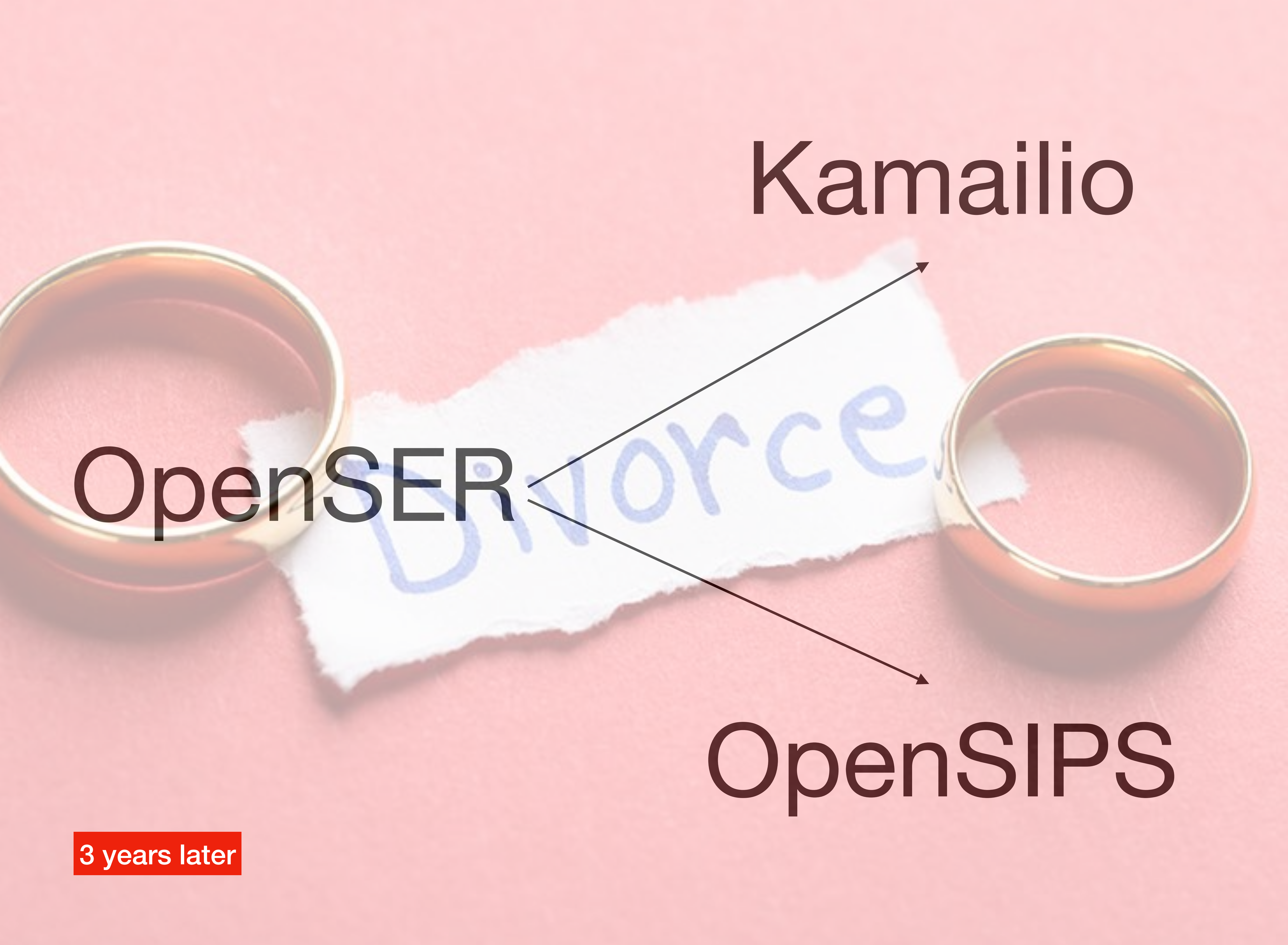
In 2004 spon-off SER as IPTEL
sold to Tekelek



SER survived as OpenSER

2005

OpenXCAP
MSRP Relay



Kamailio

OpenSER

OpenSIPS

3 years later

nat_traversal
call_control
mediaproxy_dispatcher

AG Projects

delivers and support

scalable RTC infrastructure

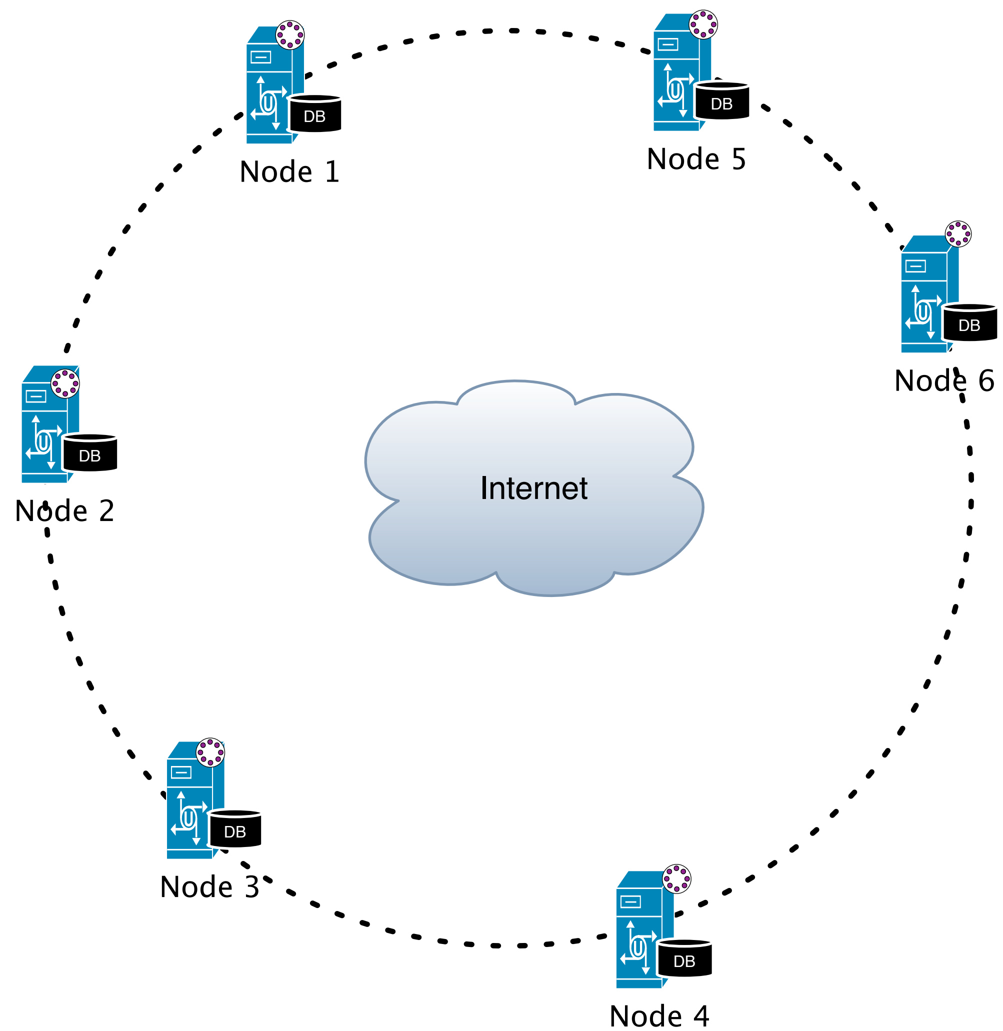
Horizontally scalable for each function

SIP Thor

Self-organizing SIP Infrastructure

Self-organizing
Horizontally scalable
Built-in disaster recovery
No single point of failure
Maintenance free

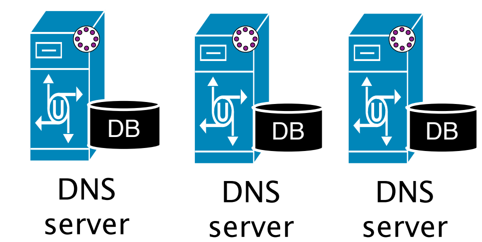
Multiple Roles
SIP Proxy/Registrar
RTP Media relay
Presence Agent
XCAP server
Voicemail server
Rating server
Provisioning server
WebRTC gateway
XMPP gateway

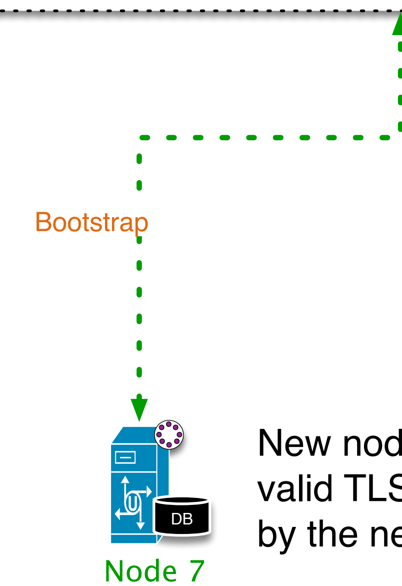


Self-provisioned DNS

Dynamic updates

_sip_udp.example.com 0 1 proxy1
_sip_udp.example.com 0 1 proxy2
_sip_udp.example.com 0 1 proxy3
_sip_udp.example.com 0 1 proxy4
_sip_udp.example.com 0 1 proxy5
_sip_udp.example.com 0 1 proxy6
_sip_udp.example.com 0 1 proxy7





New nodes are provisioned with a valid TLS certificate to be accepted by the network

All nodes are in active mode, can be distributed in multiple datacenters and any of them can handle requests from any subscriber

SIP SIMPLE client SDK (2007)

SIP SIMPLE client SDK is a Software Development Kit for development of Real Time Applications based on SIP and related protocols for media transport like Presence, Audio/Video and Instant Messaging (IM). Other media types can be easily added by using an extensible high-level API.

http://sipsimpleclient.org

Target platforms

The SDK has cross platform capabilities on Linux OS, Mac OSX and Microsoft Windows.

The library works (**with minimal changes**) on any platform that supports C and Python development environments.

Lot of hard work

Features

Non-blocking, asynchronous middleware
Including a multi-account configuration framework
TLS Security for signalling (SIP) and media (MSRP, XCAP)
Multiple media types per Session (Audio + Video + IM + File Transfer)
re-INVITE for adding and removing media to and from SIP sessions
Audio Conferencing, Presence, session based IM, HD Video
Wide-band Internet codecs: Opus, G722
Narrow-band codecs: G711, iLBC, GSM

No server can handle our TLS stack!

SIP Signalling

Session Initiation Protocol RFC3261
Session Description Protocol RFC4566
An Offer/Answer Model with SDP RFC4566

Tested in Japan

Location Discovery

Next hop address resolution based on RFC3263
(NAPTR/SRV/A DNS lookups)
Bonjour multicast DNS

Look, real DNS lookups!

NAT Traversal

SIP Signalling: Symmetric Response Routing Symmetric media RFC3581
RTP media (Audio and Video): Interactive Connectivity Establishment (ICE) RFC5245
MSRP media (IM and File Transfers): Relay Extension RFC4976 and MSRP-ACM

The good old way, 8 ports per session 2xRTP, 2xRTCP, x2 just to be sure

Voice and Video over IP

RTP, A Transport Protocol for Real-Time Applications RFC3550
Real Time Control Protocol (RTCP) attribute in Session Description Protocol RFC3605
Generation and parsing of telephone-events payload in both RTP and SDP RFC283

Instant Messaging and File Transfer

Common Presence and Instant Messaging (CPIM): RFC3862
Session Initiation Protocol (SIP) Extension for Instant Messaging RFC3428
MSRP Protocol RFC4975
Indication of Message Composition for Instant Messaging RFC3994
Message Summary Event Package RFC3842
File Transfers RFC5547
Desktop Sharing: draft-garcia-mmusic-sdp-collaboration-00 using RFB over MSRP

Multi-party Conferencing

Conference Event Package - RFC4575
A Framework for Conferencing with SIP - RFC4353
SIP Call Control - Conferencing for User Agents - RFC4579
MSRP ad-hoc multi-party chat sessions - RFC 7701

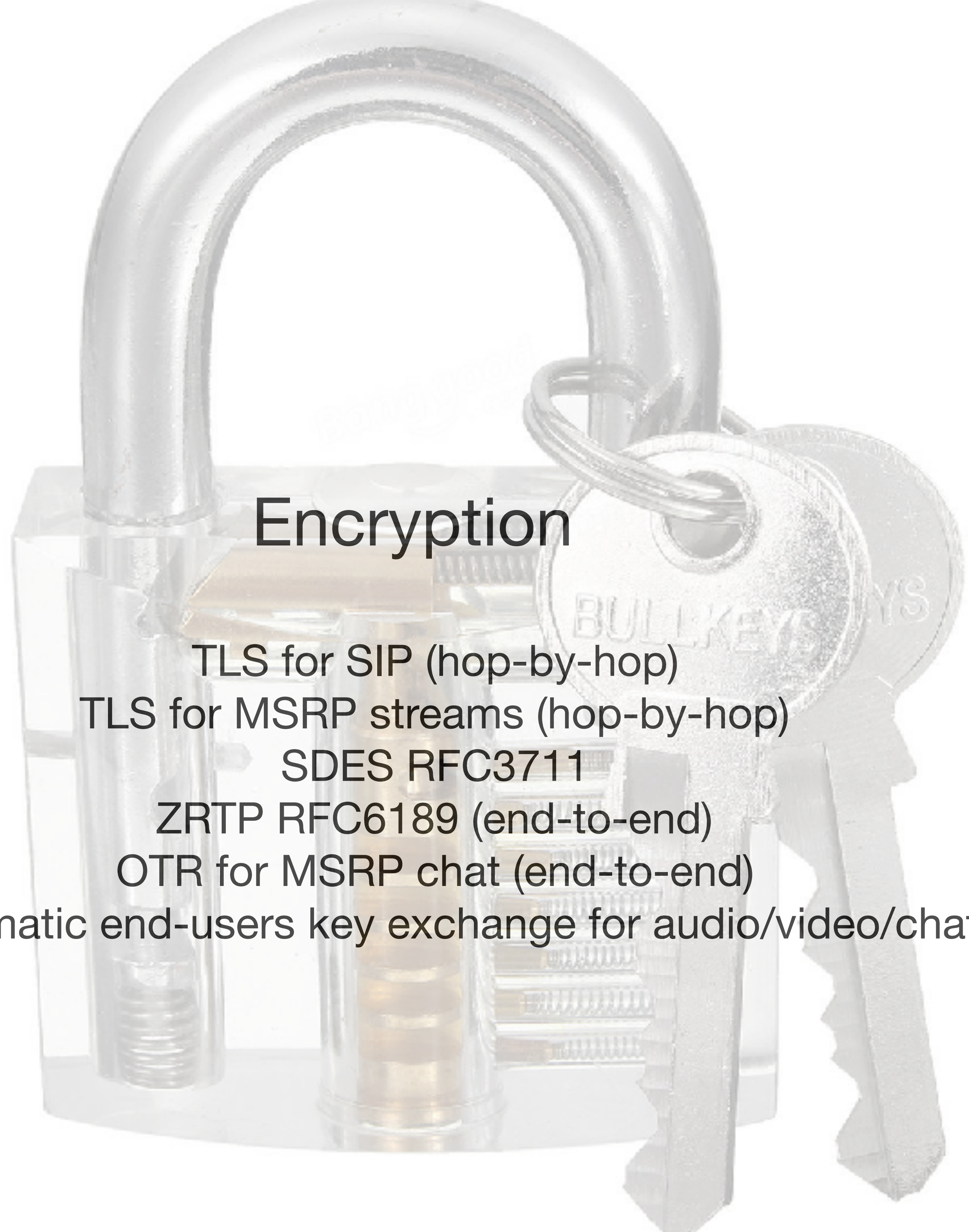
Presence

SIP Specific Event Notification RFC3265
SIP Extension for Event State Publication RFC3903
Presence Data Model (PIDF) RFC3863, RFC3379, RFC4479
Watcher-info Event Package RFC3857, RFC3858
Rich Presence Extensions to PIDF RFC4480
Contact Information Extension to PIDF RFC4482
User Agent Capability Extension to PIDF RFC5196
XCAP Protocol RFC4825
Common Policy RFC4745
Presence Rules RFC5025
Resource Lists RFC4826
RLS Services RFC4826
PIDF manipulation RFC4827

Biggest Presence farm in the world

Default policy: Yes

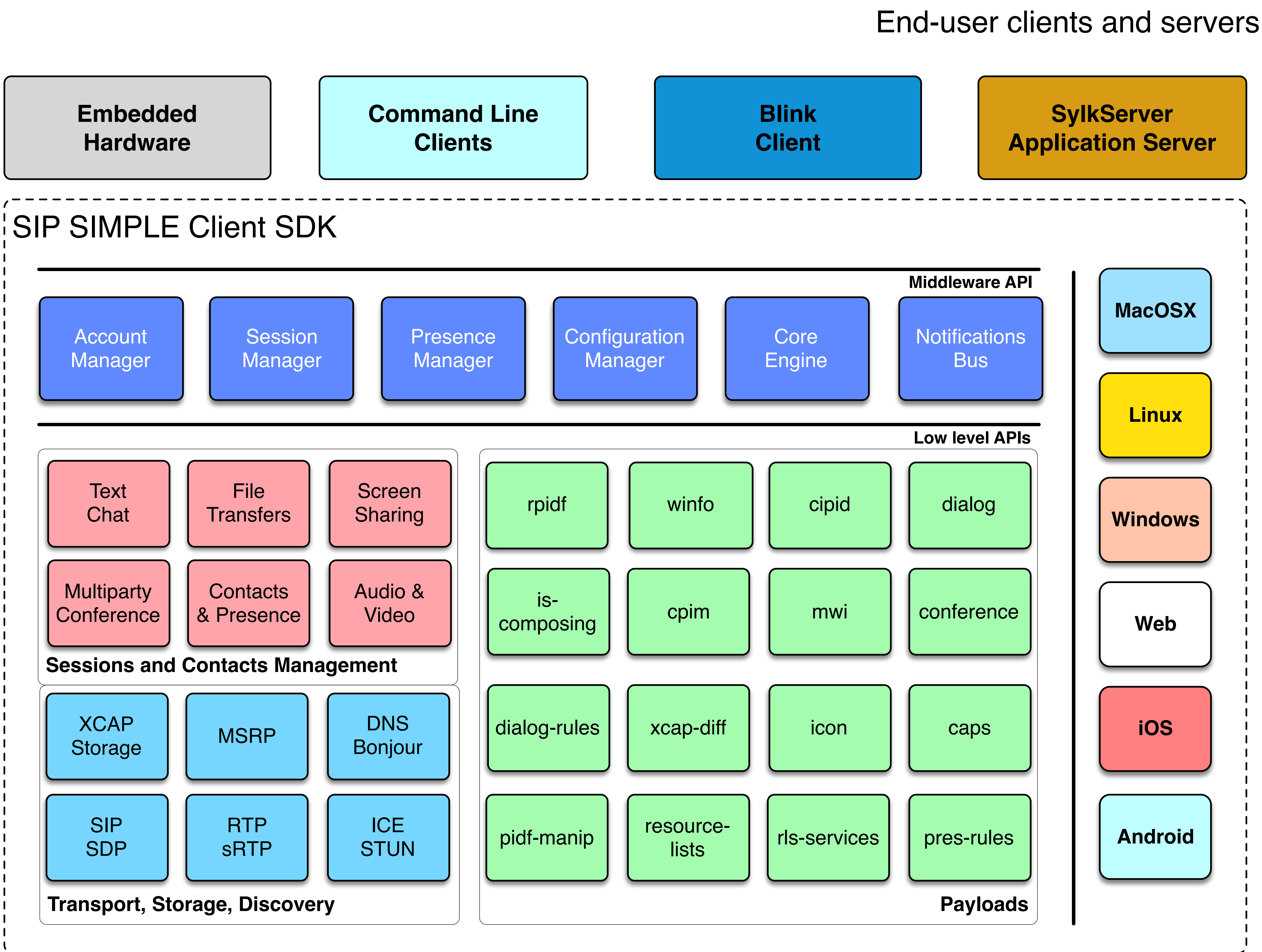
WhatsApp



Encryption

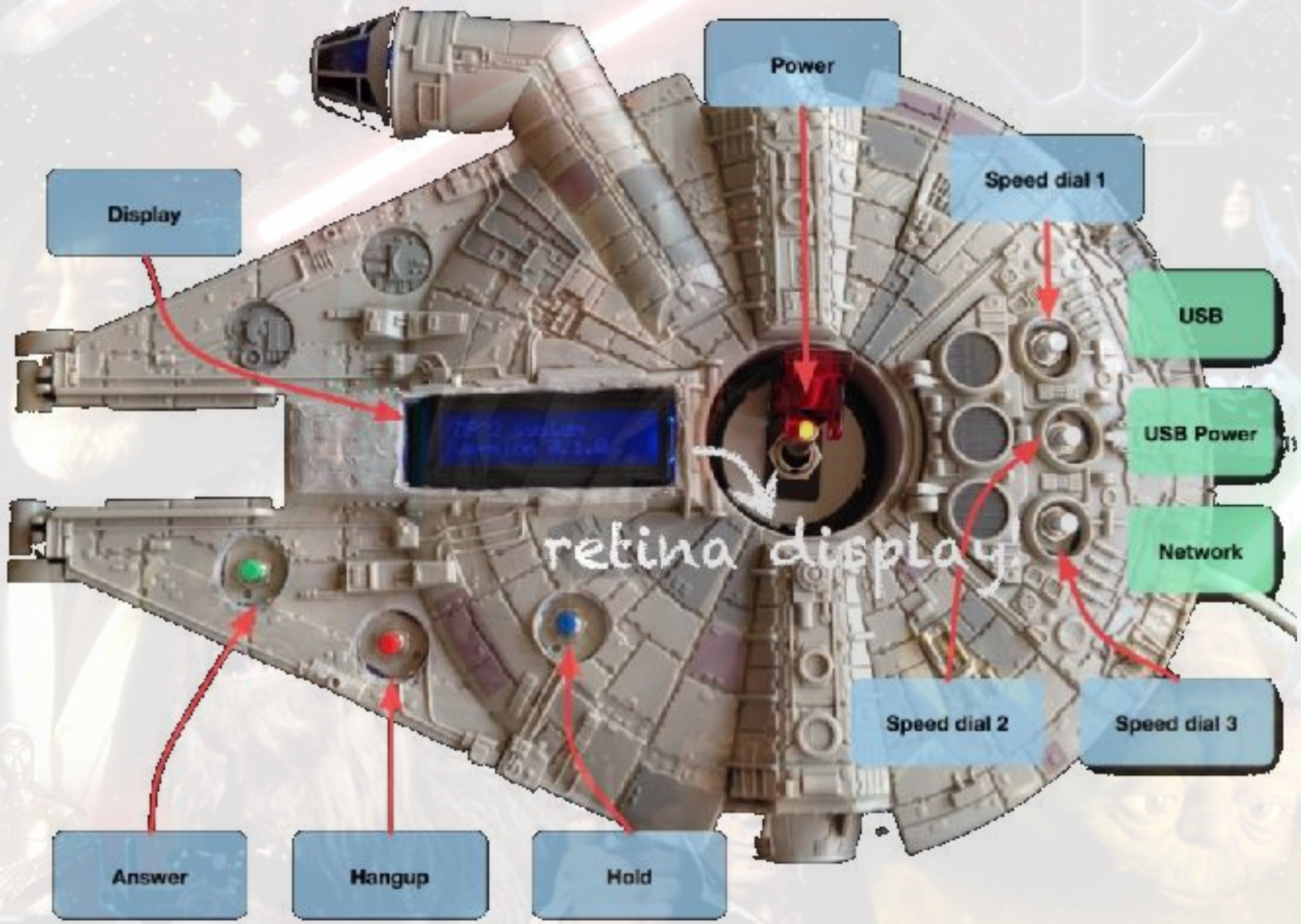
TLS for SIP (hop-by-hop)
TLS for MSRP streams (hop-by-hop)
SDES RFC3711
ZRTP RFC6189 (end-to-end)
OTR for MSRP chat (end-to-end)
Automatic end-users key exchange for audio/video/chat


Does your encryption really work?


End-user clients and servers
Embedded Hardware
Command Line Clients
Blink Client
SylkServer Application Server
SIP SIMPLE Client SDK
Middleware API
Account Manager
Session Manager
Presence Manager
Configuration Manager
Core Engine
Notifications Bus
Low level APIs
Text Chat
File Transfers
Screen Sharing
Multiparty Conference
Contacts & Presence
Audio & Video
Sessions and Contacts Management
XCAP Storage
MSRP
DNS Bonjour
SIP SDP
RTP sRTP
ICE STUN
Transport, Storage, Discovery
rpidf
winfo
cipid
dialog
is-composing
cpim
mwi
conference
dialog-rules
xcap-diff
icon
caps
pidf-manip
resource-lists
rls-services
pres-rules
Payloads
MacOSX
Linux
Windows
Web
iOS
Android

STOP TALKING AND SHOW ME THE CLIENTS!

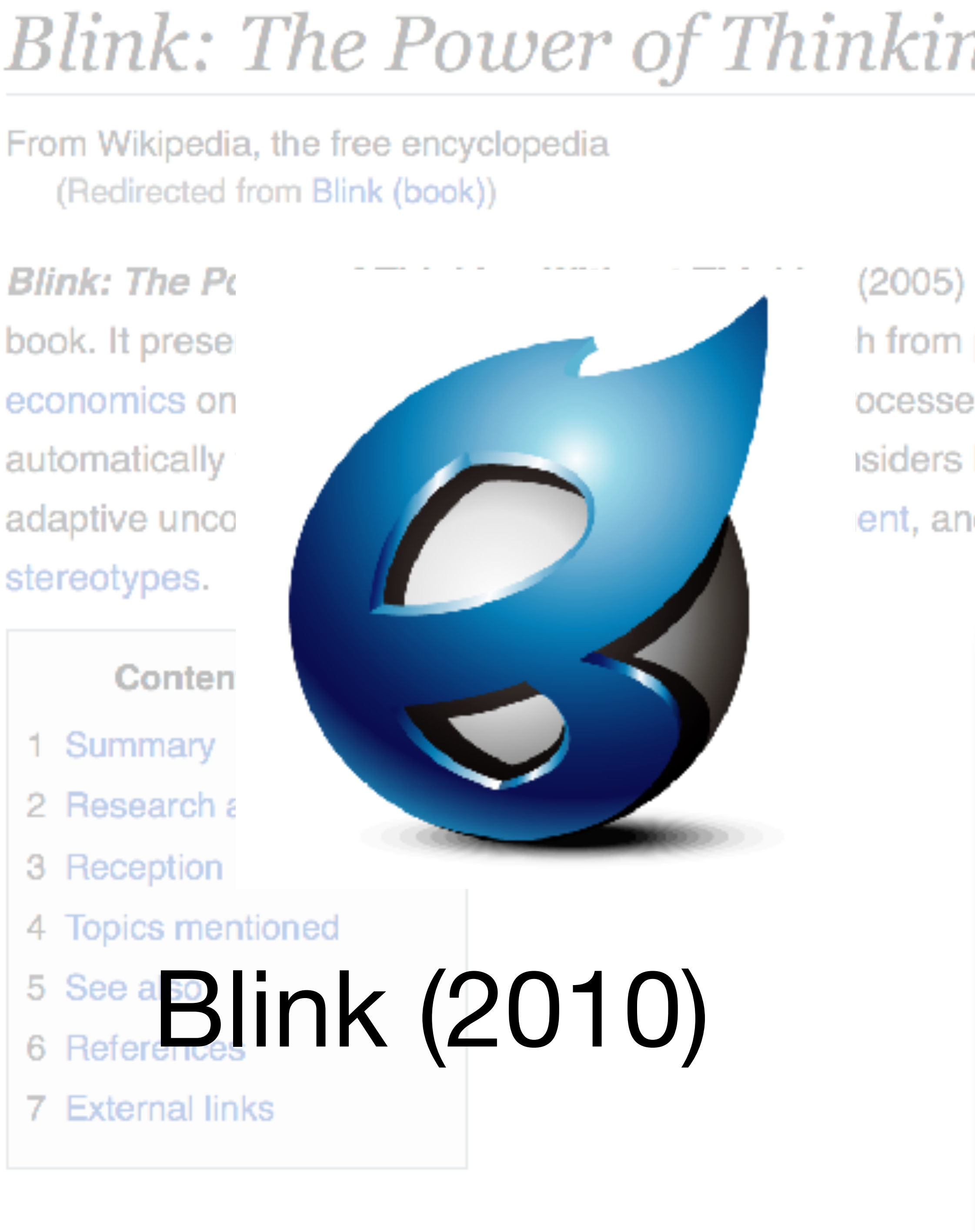
Proyecto OP^2: Open Pi Phone



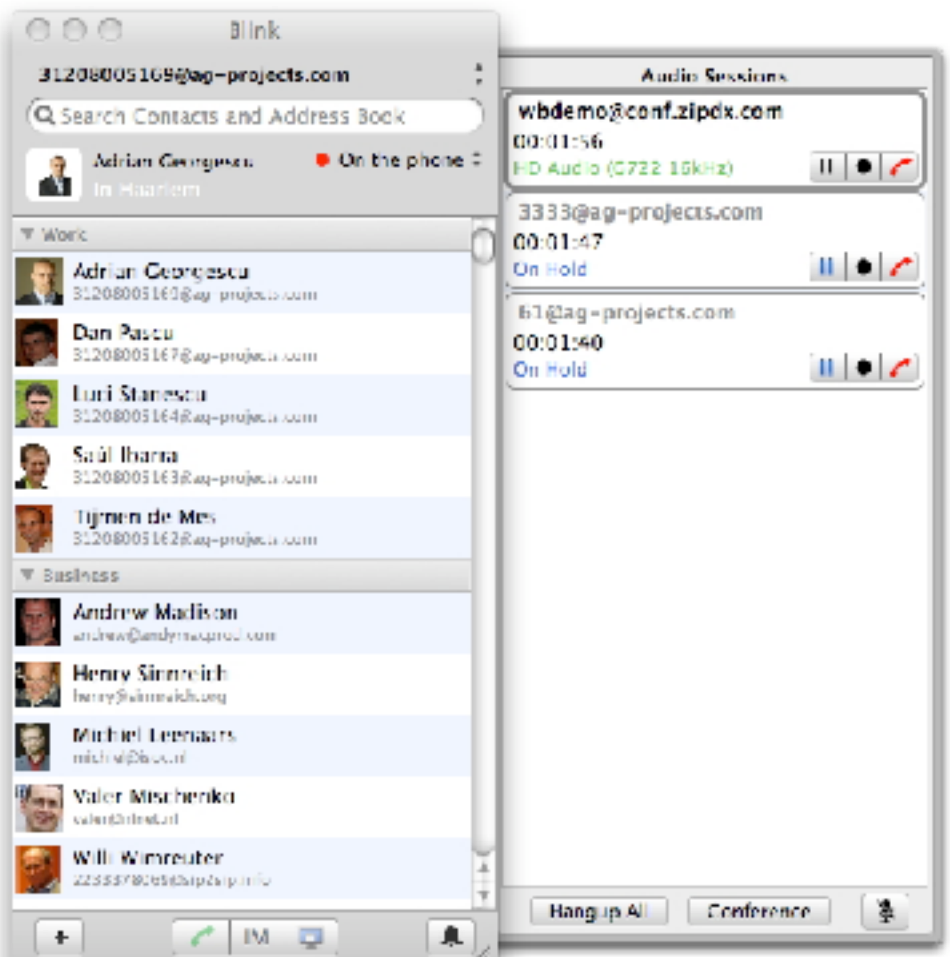
BY TIJMEN & SAUL

SIP SIMPLE Client - Command line clients

```
sip-register
sip-message
sip-session
sip-audio-session
sip-publish-presence
sip-subscribe-mwi
sip-subscribe-winfo
sip-subscribe-presence
sip-subscribe-xcap-diff
sip-subscribe-rls
```


Blink: The Power of Thinking Without Thinking
From Wikipedia, the free encyclopedia
(Redirected from Blink (book))
Blink: The P
(2005) is Malcolm Gladwell's second
book. It prese
h from psychology and behavioral
economics on
ocesses that work rapidly and
automatically
isiders both the strengths of the
adaptive unco
ent, and its pitfalls, such as
stereotypes.
Conten
1 Summary
2 Research
3 Reception
4 Topics mentioned
5 See also
6 References
7 External links
Blink: The Power of Thinking Without Thinking
ink
The Tipping Point
m Gladwell

Blink (2010)


Blink
Search Contacts and Address Book
On the phone
Work
Adrian Georgescu
Dan Pascu
Saúl Ibarra
Business
Andrew Madison
Henry Sinnreich
IM
Audio Sessions
wbdemo@conf.zipdx.com
00:01:56
3333@ag-projects.com
00:01:47
On Hold
61@ag-projects.com
00:01:40
On Hold
Hang up All
Conference


STOP THAT
THIS IS GETTING VERY SILLY
memecrunch.com


WHERE is the WEBRTC
part?


No SIP Clients
demos
for your today!

SylkServer

An extensible real-time-communications application server

SylkServer allows creation and delivery of rich multimedia applications accessed by SIP Clients, XMPP endpoints and WebRTC applications

http://sylkserver.com

SIP Signaling

TLS, TCP and UDP transports
INVITE, MESSAGE, REFER
SUBSCRIBE, NOTIFY
XMPP Gateway

Conferencing

Conference participants
Wideband RTP mixer
MSRP switch
XMPP MUC

Video conferencing

WebRTC RTP/AV
Encryption
H.264 and VP8 codecs
Opus wideband audio

NAT Traversal

SIP Outbound Proxy
ICE clients
MSRP Relay clients
MSRP ACM clients

Voice over IP

Wideband (Opus, G722)
Narrowband (G711, iLBC)
SRTP encryption (SDES, ZRTP)
Hold/Unhold

IM & Presence

MSRP and XCAP protocols
CPIM envelope
Is-composing indicator
Delivery reports

File Transfer

MSRP protocol
Progress reports
Conference-info extension
Conference room persistent

Gateways

SIP/XMPP chat/audio
XMPP/SIMPLE presence
SIP/WebRTC RTP/AV
IRC chat

Sylk SIP Conference

SylkServer allows SIP end-points to create ad-hoc conference rooms by sending INVITE with SDP content to a random username at the hostname or domain where the server runs. Other participants can then join by sending an INVITE to the same SIP URI used to create the room.

Multiparty Chat RFC 7701
Conference Info RFC4575
RTP Audio
File transfers and Screen sharing
No Video

Sylk XMPP gateway - RFC 7248

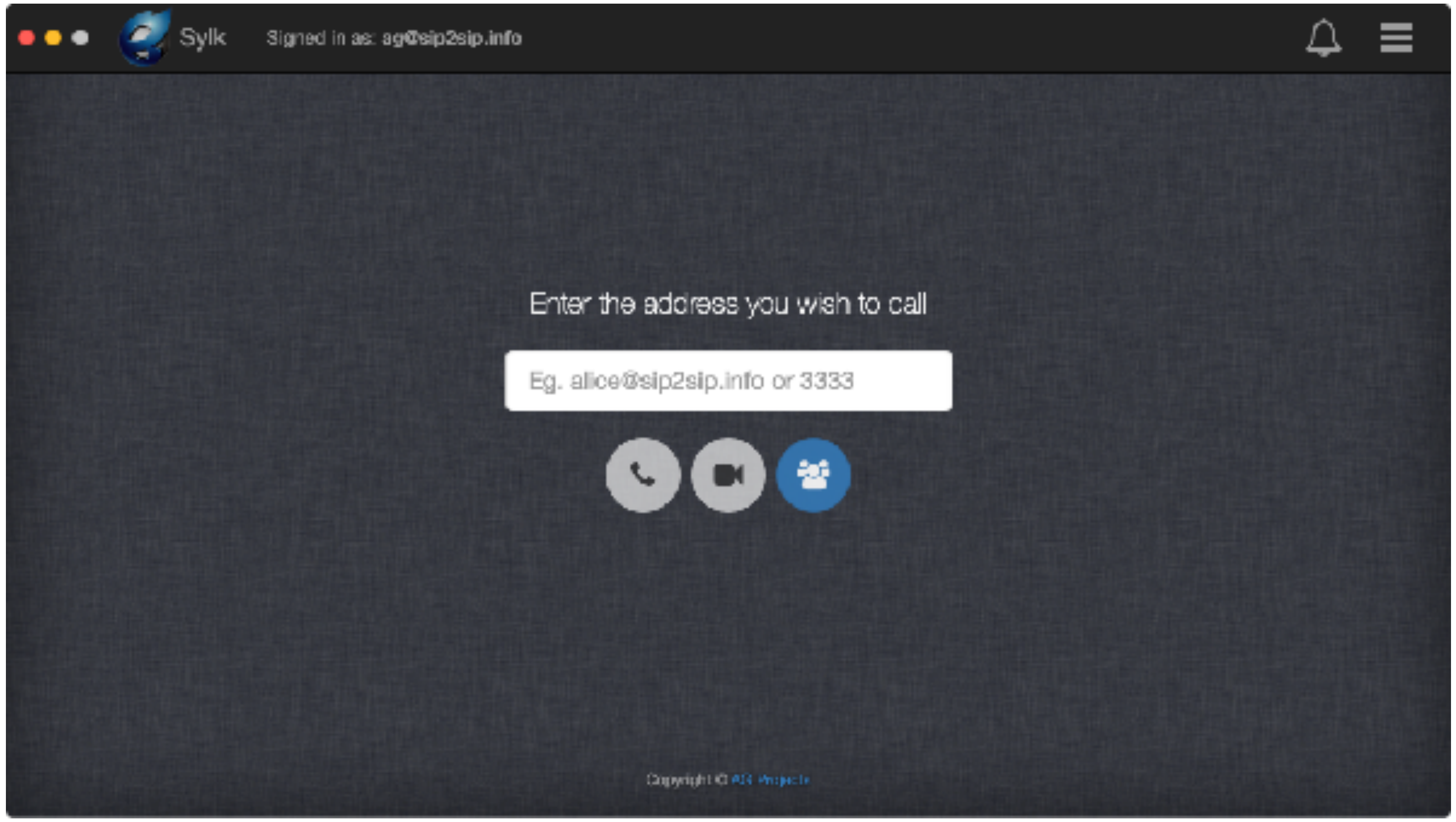
SylkServer can act as a transparent inter-domain gateway between SIP and XMPP protocols. This can be used by a SIP service provider to bridge out to external XMPP domains or to receive incoming chat messages and Jingle audio sessions from remote XMPP domains to its local SIP users. In a similar fashion, a XMPP service provider can use the gateway to bridge out to external SIP domains and handle incoming chat requests from SIP domains to the XMPP users it serves.

See http://sip2sip.info for a working example.

Sylk WebRTC gateway

Used to bridge audio and video calls between SIP clients and WebRTC applications. This application supports transparently any audio/video codec negotiated by the end-points, however WebRTC has standardised particular codecs for the use on the web, therefore the SIP clients must support the same set of codecs.

See https://webrtc.sipthor.net for a working example.

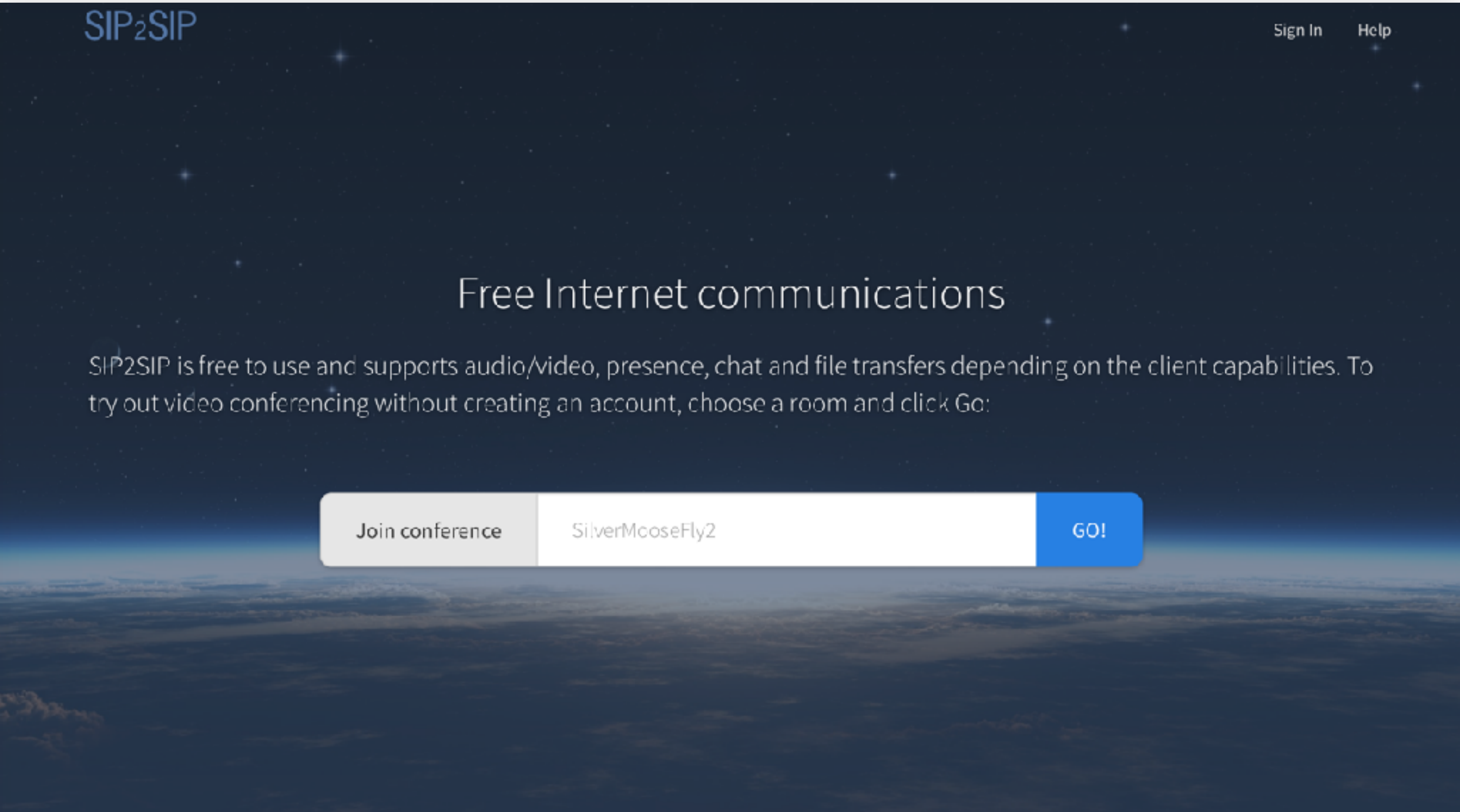


Sylk Client

Web site: https://webrtc.sipthor.net
Electron app for desktop: Sylk WebRTC Client
iOS and Android (preview beta): Blink Mobile

SIP2SIP.info

Free SIP accounts


SIP2SIP
Sign In
Help
Free Internet communications
SIP2SIP is free to use and supports audio/video, presence, chat and file transfers depending on the client capabilities. To try out video conferencing without creating an account, choose a room and click Go:
Join conference
SilverMooseFly2
GO!

Sylk WebRTC conference

This application allows WebRTC enabled end-points to organise ad-hoc audio and video conferences. SylkServer implements Selective Forwarding Unit (SFU) functionality **by using Janus backend**. SFUs use little resources on the server side, allowing for handling much more load than classic MCUs.

The bandwidth usage is optimised in such a way that independent of the number of participants present in the conference, the bandwidth required by each participant is not greater than for a direct video call between only two participants.

Horizontally scalable with SIP Thor

Sylk Suite

Sylk Server + Sylk Client + API

SIP + XMPP + WebRTC
for operators with SIP Infrastructure

apt-get install sylkserver

Zero configuration!

Really, how about TLS?

Roadmap

~~Multiparty Video~~
~~Multiparty Screen sharing~~
Multiparty file sharing
Multiparty text chat
Interoperability with non-webrtc audio (SIP)
Interoperability with SIP/MSRP and XMPP MUC

Sponsored by NLNet

github.com/agprojects

AG Projects repositories

@agprojects

info@ag-projects.com