


TLS for SIP and RTP

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v2.0



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Why TLS?



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Why TLS?

- Authenticity (man-in-the-middle)
- WebRTC requirement
- Authentication... sort of
- Privacy
 - Metadata
 - DTMF



SIP

- SIP is a plain text protocol

```
INVITE sip:alice@example.com SIP/2.0
Via: SIP/2.0/UDP 192.0.2.110:5060;branch=z9hG4bK742a0872;rport
Max-Forwards: 70
From: <sip:bob@example.com>;tag=as30c330ef
To: <sip:alice@example.com>
Contact: <sip:bob@192.0.2.110:40958>
Call-ID: 2f1496093878509da1074b153b880f9@192.0.2.110:5060
Cseq: 102 INVITE
User-Agent: Some Device v1.0
```



SIP

- SIP is multi-protocol
 - UDP is the original. Has issues with datagram size overflow. Retransmission due to loss.
 - TCP seems like an odd choice, but adds reliability. Plays nicer with NAT.
 - SCTP is a hybrid. Datagram based, with 4-way handshake. Not widely supported.



SIP

- SIP is signaling only
 - Call setup, tear down, progress
 - Messaging, routing
 - No media
 - Session Description Protocol ties to the next protocol, typically Realtime Protocol



SIP

- SIP has NAT traversal issues
 - NAT sucks
 - IPs and ports are embedded in SIP, SDP
 - Keep-alives are required
 - Some devices try to “help” by being “smarter” than the endpoints (aka ALG). These are evil.
 - NAT sucks and should die in a fire



SDP

- SDP is an offer/accept protocol

```
v=0
o=root 779795551 779795551 IN IP4 192.0.2.1
s=Asterisk PBX 13.13.13
c=IN IP4 192.0.2.1
t=0 0
m=audio 10108 RTP/SAVP 9 0 18 101
a=rtpmap:9 G722/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
a=sendrecv
```



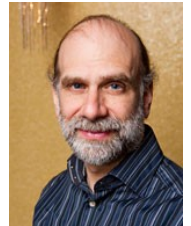
RTP

- Realtime Protocol
 - Independent protocol from SIP
 - Supports wide variety of media types, e.g. audio, video, DTMF
 - Also used by H.323, others
 - Has a companion protocol named RTCP which gives feedback reports to the far side



TLS

- TLS started life as SSL
- Developed at Netscape to secure HTTP
- SSLv2 (1995) had some major flaws
 - “Anyone, from the most clueless amateur to the best cryptographer, can create an algorithm that he himself can't break” -Bruce Schneier
- SSLv3 released in 1996



TLS

- SSL was turned over to the IETF, became TLS
- Version 1.0 released in 1999, 1.1 in 2006, 1.2 in 2008
- TLS used to downgrade itself to SSL, but this is now considered harmful
- Surprisingly, many sites still use SSLv2



Certificate Authorities

- Biggest problem with encryption is key management
- TLS uses Certificate Authorities (CA)
- Certificate Authorities
 - Sign certificates
 - Are supposed to verify before issuing
 - Can revoke certificates



Certificate Authorities

- Client ship with many well-known CA public keys
- Anybody can create their own CA. This can be used in an enterprise, for example.
- Or, skip the whole thing and have a certificate sign itself



TLS + SIP

- SIP over TLS requires either TCP or SCTP
- Simple TLS wrapper around the SIP protocol, just like HTTPS
- No STARTTLS



TLS + RTP

- RTP data can be optionally encrypted
- Encryption is easy. Hard part is key negotiation.



TLS + RTP

- RTP data can be optionally encrypted
- Encryption is easy. Hard part is key negotiation. 4 options:
 - SDES
 - DTLS
 - MIKEY
 - ZRTP



SDES

- Key is sent in plain text by presumed secure channel
- Fantastically simple
- Each SIP hop must independently negotiate key, do encrypt/decrypt
- No end-to-end security

a=crypto:1 AES_CM_128_HMAC_SHA1_80
inline:JmrRpWso6K1EBGw9Q3ua+nTbM86f+0WYZI25fC3x



DTLS

- Datagram TLS
- Modifications of TLS to make it datagram friendly
- Adapts to packet loss, out of order delivery
- 4-way handshake with cookie
- Stateless from datagram to datagram
- Keying is end-to-end



DTLS Example

- OpenSSL can do DTLS

```
$ openssl s_server -dtls1 -accept 5555 -cert server.pem -key server.key  
Using default temp DH parameters  
Using default temp ECDH parameters  
ACCEPT
```

```
$ openssl s_client -dtls1 -connect localhost:5555  
CONNECTED(00000003)
```



MIKEY

- I don't know anything about MIKEY, thus it's not worth knowing about

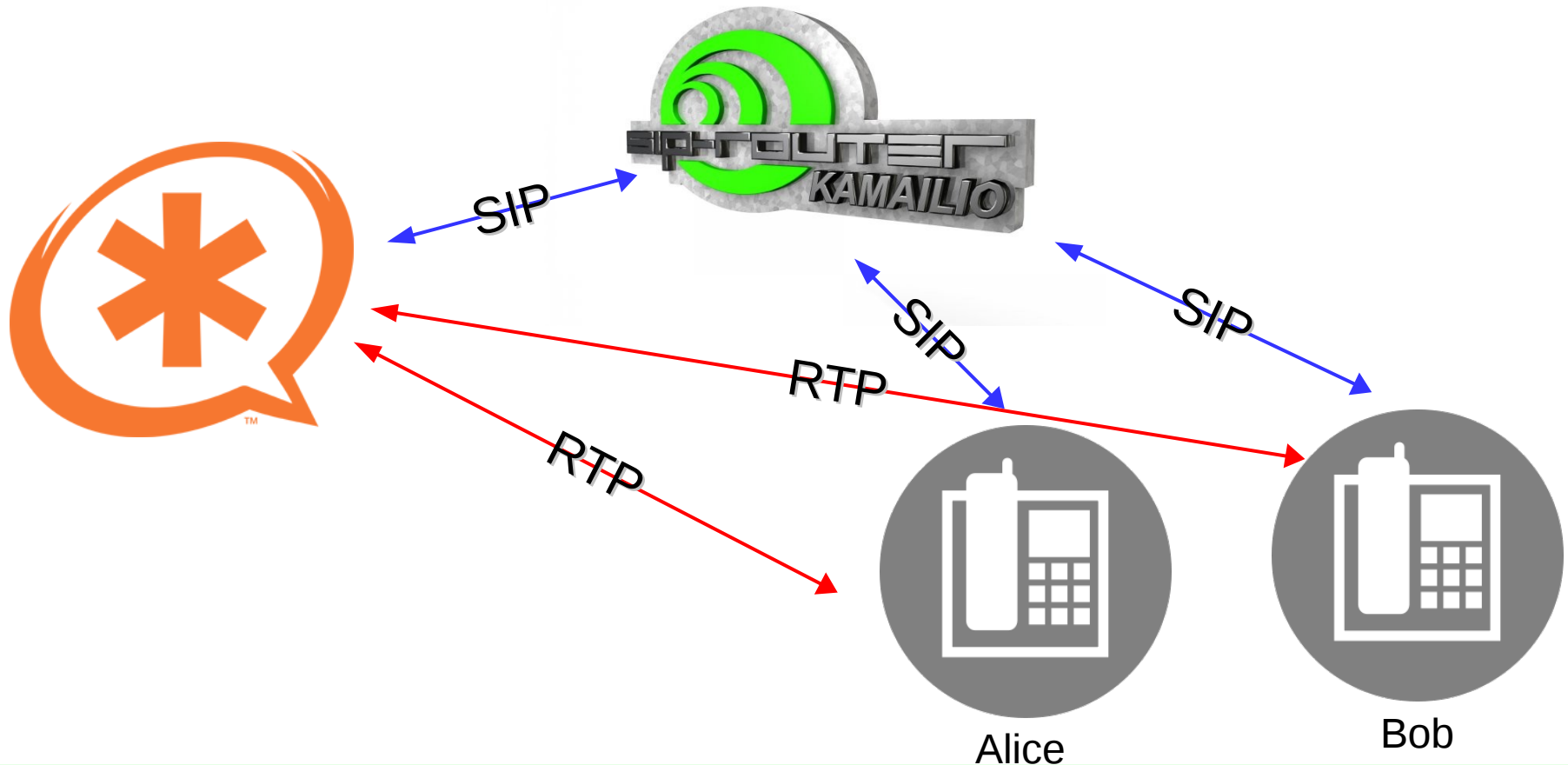


ZRTP

- Developed by Phil Zimmerman of PGP
- End-to-end opportunistic encryption
- Requires no certificates, authorities
- Vulnerable to man-in-the-middle, but provides authentication hash
- Hash cache increases security over time



Secure VoIP Architecture



Create a Certificate

- `openssl req -out certreq.csr -new -nodes -days 9999 -keyout key.pem`
- Certificate signing request (CSR) stored in “certreq.csr”
- Private key in “key.pem”
- Have CSR signed by CA, or create your own
- `openssl x509 -req -days 9999 -in certreq.csr -signkey key.pem -out cert.pem`



Asterisk

- `cat key.pem >>cert.pem`
- `sip.conf`

```
[general]  
tlsenable=yes  
tlscertfile=/path/to/your/cert.pem  
tlscacfile=/path/to/your/ca.pem  
;tlsbindaddr=0.0.0.0:5071  
;bindport=5070
```



Asterisk

- sip.conf

[peer]

...

transport=tls

encryption=yes



Kamailio

- kamailio.cfg

```
enable_tls=yes  
loadmodule "tls.so"  
modparam("tls", "config", "/etc/kamailio/tls.cfg")  
  
;listen=tls:0.0.0.0:5061
```



Kamailio

- `tls.cfg`

```
method = TLSv1
verify_certificate = no
require_certificate = no
private_key = /etc/kamailio/key.pem
certificate = /etc/kamailio/cert.pem
ca_list = /etc/kamailio/ca.pem
```

```
[client:default]
```

```
verify_certificate = yes
require_certificate = yes
```



Jitsi

- Account
 - SIP ID
openwest4XX@ip.address
 - Password soopersecret
- Connection
 - Registrar ip.address
 - Port 5061
- Proxy ip.address
- Port 5061
- Preferred Transport
TLS
- Security
 - SDES
 - RTP/SAVP indication
MandatoryClick to add
Text




Limitations

- Gateways can switch to unencrypted
 - sips: URI scheme conveys desire that the entire stream should be encrypted. YMMV.
- PSTN will probably never be encrypted
- Keying protocols are a headache
- Encryption adds latency



The End


Questions?

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