

Skype and the Flux-Capacitor

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Skype Overview

- „It just works“: no problems with firewalls
- Skype makers known for Spyware-contaminated filesharing software ([KazaA](#))
- built-in software-update function in Skype
- Doesn't adhere to any standards – no third-party offers
- Closed source – Open Source Skype would be nice and this may be within reach now
 - Whom to entrust with your phone calls?
 - Encryption: Who owns the keys???



VoIP Introduction

- Voice over Internet Protocol (and often also Video)
- Distinction: call signalling and content (voice/video)
- Signalling Protocols e. g. SIP, H.323, ...
- Content transmission e. g. RTP (Realtime Transfer Protocol)
- Quality: Low latency, low Jitter, low packet loss
- Open: Success of VoIP depends on standards: we want interconnection of different vendor solutions
- If signalling and content are sent via different path we may get problems with firewalls



VoIP Signalling Protocols

- IAX: Inter Asterisk Exchange: use same path for signalling and content – firewall-friendly
- SIP: Session Initiation Protocol
- Jabber/Jingle (Google Talk)
- H.323: ITU-T Standard
- MGCP: Media Gateway Control Protocol
- SCCP/Skinny (proprietary CISCO)
- UNISTIM (proprietary Nortel)
- Skype (proprietary): optionally one signalling path
→ „Asterisk is first PBX in history to natively support proprietary IP terminals from the two biggest players in VoIP“ [[MSM05](#)]



VoIP: Bunch of Firewall Problems

- SIP, Jingle, H.323 are signalling protocols
→ only for call-management
- For Content: RTP (Realtime Transfer Protocol)
RTP-Port is dynamically negotiated
- Firewall (NAT!) problems
 - Remedy (or workaround): NAT-Traversal protocols, or intelligent firewall
 - For Linux/Netfilter: SIP+H.323 connection tracking and NAT modules
 - or use IAX!



Standards for Encrypted Telephony

- **encryption option** for IAX with pre-shared keys
- VoIP can be routed via a VPN, e. g. **OpenVPN**
- **SRTP/SRTCP**: needs a key management scheme Master-Key: call parties need to agree on a key
- Phil Zimmermanns **ZRTP**: SRTP + Diffie-Hellman
 - call parties need not know each other!
 - uses existing SIP infrastructure
 - approved as **informational RFC**
- **IETF DTLS** based on certificates for negotiating SRTP key – will become Standards-Track RFC (!)

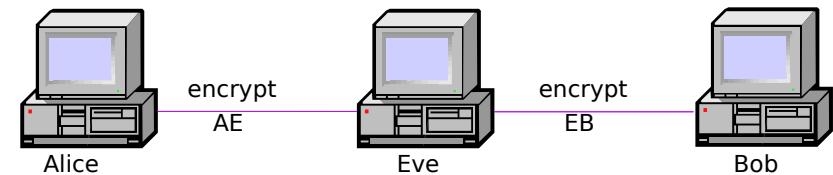


Security with Encrypted Telephony

- Traffic Analysis: „Who with whom“
→ nearly conforming to EU Data Retention :-)
- IAX transmits session information in clear
- Even when using a VPN we can determine that there is VoIP-content due to packet frequencies and -sizes
- SRTP without ZRTP, IAX or VPN:
needs prior agreement of calling/called parties
- ZRTP **MITM**-Protection: call partners can read key-hash to each other – hard to fake by man in the middle



Excursus: Man in the Middle



- Eve – the woman in the middle – does key exchange with Alice and Bob
- There are two keys, AE und EB
- But: Alice and Bob can compare key checksums by reading them to each other over the phone
→ Authentication via speech!



Existing Implementations

General:

- IAX encryption: no personal experience
- OpenVPN + Asterisk IAX (or SIP/RTP) works well

ZRTP:

- SRTP/ZRTP not yet in Asterisk but ZFone can be used
- Phil Zimmermann's **ZFone** as add-on to existing RTP-based software
- GNU **ccRTP** supports ZRTP
- GPL-Client **Twinkle** with ZRTP



Alternatives to Skype

- SIP-Phones (Ekiga, Twinkle, Linphone)
- Chat: multiprotocol-client Pidgin
- Chat + telephony: Google Talk (Jabber extension „Jingle“ for Voice/Video)
- Open Source telephony: Asterisk: SIP, Jabber/Jingle, and lots of other protocols ...
- Encryption: Phil Zimmermann's ZRTP + SRTP or IETF's DTLS (certificate-based) + SRTP



History of Publications about Skype

- Biondi, Desclaux: Silver Needle in the Skype, Black-hat Europe 2006 [BD06]
- Desclaux, Kortchinsky: Vanilla Skype [DK06a, DK06b], Recon 2006
- Biondi, Desclaux, Kortchinsky analyzed the network obfuscation and compression algorithms of Skype but didn't publish their work
- they called the network obfuscation function „Flux Capacitor“
- 2010-07-07 Sean O'Neil:
[Skype's Biggest Secret Revealed](#)



Skype Security Considerations

- Writeable text-segment:
Decrypts itself when called
- could load code over the network into memory
- when asking for updates Skype sends its [Skype-ID](#)
 - maybe this is used for sending you your own personalized (trojan?) update?
- Skype can find out IPs of other Skype clients [BD06, p. 61] – there is already a [service](#) announced to discover Skype IPs by username
 - imagine the possibilities when coupled with Geo-location



Skype Security Considerations

- Remote Scan: We can make another node connect to a specified IP-Adress – even in the network perimeter (inside Firewall) of that host [BD06, p. 82]
- Censor API for chat activated for chinese Skype calls ContentFilter.exe – API present in all Skype versions [DK06b, p. 40]
- AP2AP protocol for communication of non-Skype applications via Skype [DK06b, p. 42]
 - Information Exfiltration (file transfer)
 - remote control
- ⇒ that's why I chose [not to use Skype](#)



Skype Network Obfuscation: Flux Capacitor

- A big obfuscated function first described by Desclaux/Kortchinsky: „it is not an improvement of the flux capacitor“ [DK06a, p. 52]
- they even describe a method how to generate C-code from the binary [DK06a, p. 55–65]
- obfuscation function takes a 32-bit seed and computes an RC-4 key
- the RC4-key is used to encrypt the UDP-packet or TCP-stream
- for TCP the seed is sent in clear as the first bytes of the stream



Skype Network Obfuscation: Flux Capacitor

- for UDP the CRC32 of source-IP, dest-IP and per packet ID XOR an initialisation vector (IV) is used as the seed
- The packet ID and IV are in clear in every packet
- ... so everything is known about the RC-4 key
- It's just an obfuscation layer, no real encryption because the key is known (security by obscurity)
- But the obfuscation is so "good" that it hasn't been published until now
- I've been able to verify this for TCP, and now also for UDP



Excursus: CRC32

- Silver Needle suggests UDP uses CRC32 – there is example code from Vanilla which has a CRC32
- The well-known crc32 from Ethernet, ZIP, etc. uses the same polynomial but a different finalisation:

```
def zlib_crc (s, seed = 0) :
    state = seed ^ 0xFFFFFFFF
    state = do_crc (s, state)
    return state ^ 0xFFFFFFFF

def skype_crc (s, seed = 0xFFFFFFFF) :
    return do_crc (s, seed)
```



Excursus: CRC32

- zlib crc32 uses initial value 0, skype uses 0xFFFFFFFF
→ effect is the same internal state
 - ⇒ So we can use crc32 from zlib for skype crc:
- ```
def skype_crc (s, seed = 0xFFFFFFFF) :
 mask = 0xFFFFFFFF
 return (crc32 (s, seed ^ mask)) ^ mask
```
- this implementation allows part-wise computation:
- ```
x = skype_crc ('abcd')
x = skype_crc ('efgh', x)
y = skype_crc ('abcdefg')
assert (x == y)
```



Skype Network Obfuscation: Compression

- Before encryption Skype uses an undocumented arithmetic compression algorithm [DK06b, p.7–8]
- Arithmetic compression close to Huffman but using reals
- Sean O'Neil promises to publish the compression, too: "The compression algorithm required to complete the decoding of Skype packets will be published in December this year at 27C3"



Skype and Cryptography

- not everything is just obfuscation
- Skype uses RSA keys signed by Skype master RSA key [DK06b, p. 12-18] Skype public keys are compiled into binary
- uses shared secret: MD5-hash of user information
<login>\nskyper\n<password>
- Skype trusts Skype master RSA key
- local RSA key used for session key exchange
- when searching for a users you get their public key
- clients authenticate to each other by signing 8-byte challenge with their RSA key



Skype Communication Security

- Voice/Video calls are encrypted with session key
- But: Skype voice engine can generate encrypted trace files if certain variables are set [DK06b, p. 56]
- XOR with 31 byte key – probably breakable
- +AP2AP protocol this may permit eavesdropping

Kurt Sauer, „Leiter der Sicherheitsabteilung von Skype“ im [ZDNET Interview](#): „Wir stellen eine sichere Kommunikationsmöglichkeit zur Verfügung. Ich werde Ihnen nicht sagen, ob wir dabei zuhören können oder nicht.“



Debugging Skype

- All sorts of debugger checks
- but under Linux we can debug with GDB
- ... as long as we don't set more than 2 breakpoints (HW)
- Nice: Wait for Skype to call into lib:
`break __libc_start_main`
- Answer y to “Make breakpoint pending on future shared library load?”
- In that state skype has decrypted its text-segment



Skype Task-Force?

- Check how to decrypt UDP packets
- implement compression protocol or wait for 27C3
- Use that to implement a Wireshark dissector for Skype
- Chat
- ... Voice?
- ...



Bibliography

- [BD06] Philippe Biondi and Fabrice Desclaux. Silver needle in the skype. In *Blackhat Europe*, Netherlands, March 2006.
- [DK06a] Fabrice Desclaux and Kostya Kortchinsky. Vanilla skype part 1. In *Recon*, Montreal, Canada, June 2006.
- [DK06b] Fabrice Desclaux and Kostya Kortchinsky. Vanilla skype part 2. In *Recon*, Montreal, Canada, June 2006.



Bibliography

- [MSM05] Jim Van Meggelen, Jared Smith, and Leif Madsen. *Asterisk – The Future of Telephony*. O'Reilly, September 2005.