```
void fuzz(char* buf, int& len){

    int q = rand()%20;

    if (q == 7){
        int ind = rand()%len;
        buf[ind] = rand();
    }

    if(q == 5){
        for(int i = 0; i < len; i++)
            buf[i] = rand();
    }

    if(q == 11){
        int l = rand()% MAX_PACKET_LEN;
        *len = l;
    }
}
```

Adventures in Video Conferencing

About Me

- Natalie Silvanovich AKA natashenka
- Project Zero member
- Previously did mobile security on Android and BlackBerry
- Defensive-turned-offensive researcher

Video Conferencing

- Video conferencing has expanded greatly in the past 5 years
  - Browsers
  - FaceTime
  - WhatsApp
  - Facebook
  - Signal

WebRTC

What is WebRTC?

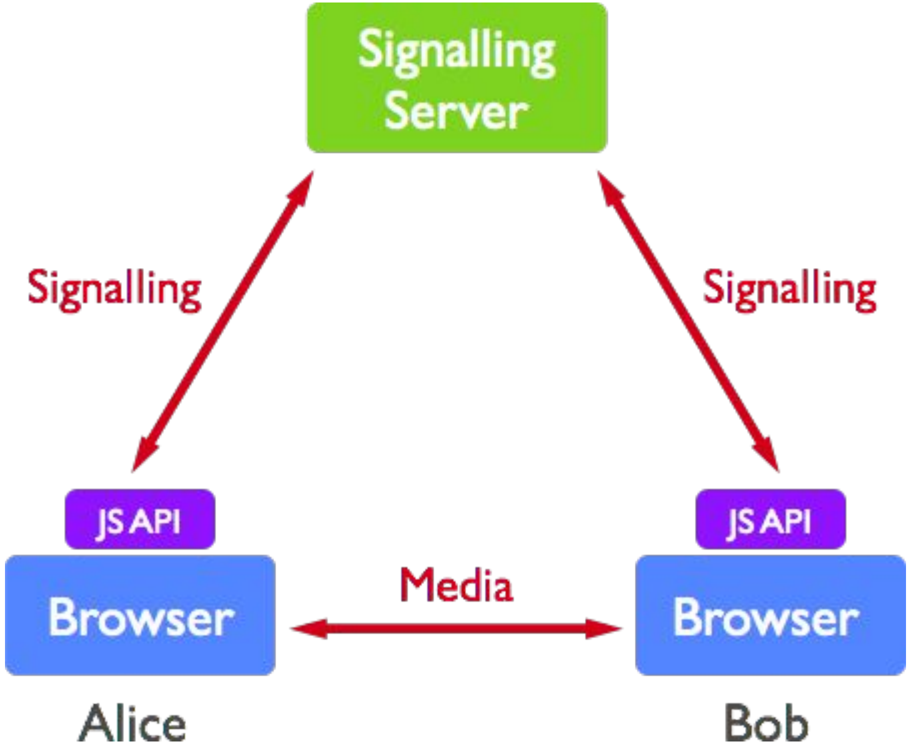
- RTC = Real Time Communication
- Audio and video conferencing library maintained by Chrome
- Used by
  - Browsers (Chrome, Firefox, Safari)
  - Messaging applications (Whatsapp, Facebook Messenger, Signal, SnapChat, Slack, etc.)
- Little security information available

Known Security Bugs

List of Known Security Bugs

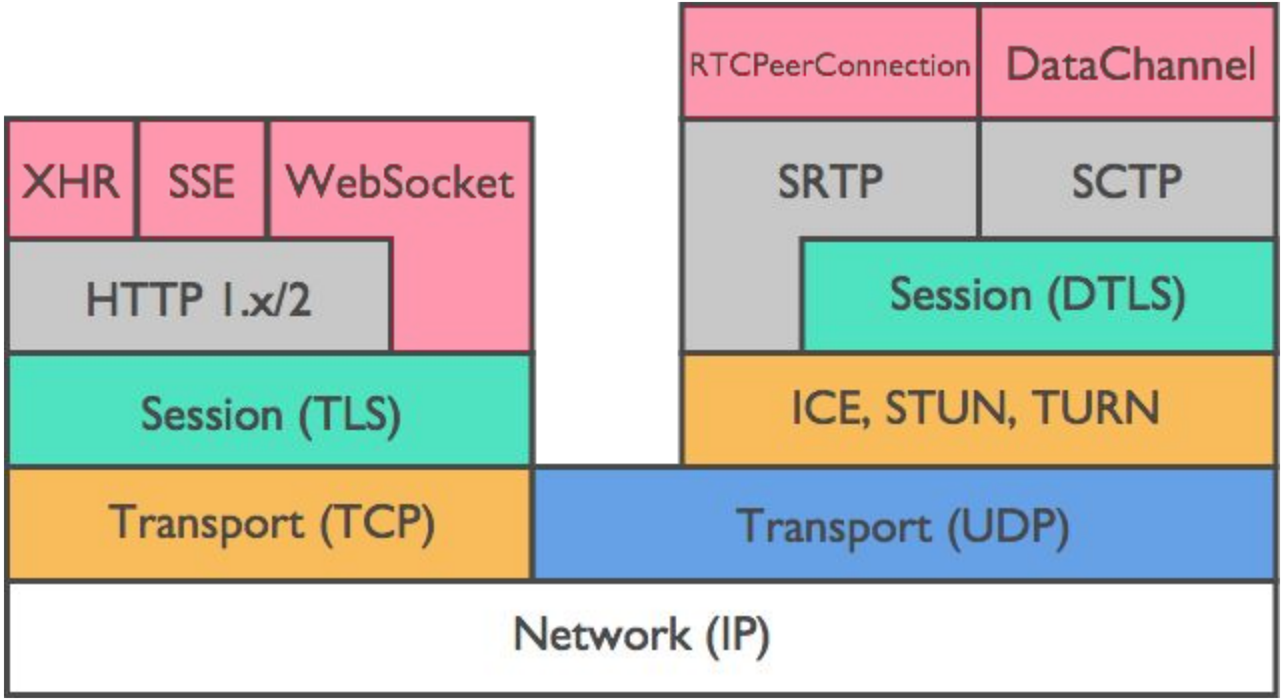
- (None so far)

WebRTC Architecture

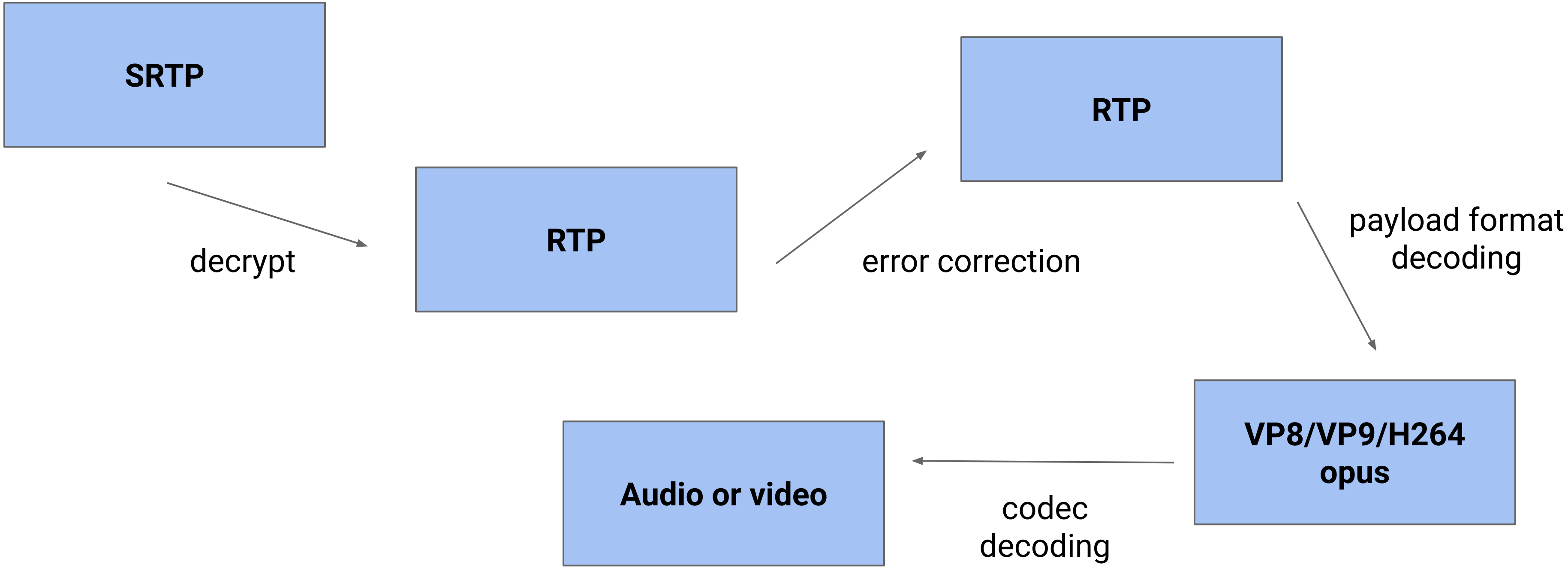


Google

WebRTC Architecture



Packet Decoding Sequence



Idea 1: Session Description Protocol

- SDP is the most sensitive interface of WebRTC
  - WebRTC requires parsing untrusted SDP with no user interaction
- Used WebRTC library to create SDP fuzzer on commandline
- Reviewed SDP code
- No bugs!
- Some platforms implement separately

Idea 2: RTP and Media Protocols

- WebRTC has already implemented fuzzers for RTP, media protocols and codecs
  - But what about end-to-end?
- Wrote end-to-end fuzzer for RTP

Evolution of a fuzzer

Prototype

- Altered Chrome to add fuzzer
- Had one browser instance ‘call’ another
- Crashed roughly every 30 seconds
- Learned that the concept would generally work
- Got very shallow bugs that blocked fuzzing fixed

Evolution of a fuzzer

Client Fuzzer

- Wrote C++ client that interacts with browser
  - Lighter weight than browser
  - Can run against any target
  - Pro: crashes are guaranteed to work on browser
  - Con: slow
- Found additional end-to-end vulnerabilities in WebRTC

Evolution of a fuzzer

Distributed Fuzzer

- Wrote command line RTP emulator with help of WebRTC team
  - Pro: extremely fast, runs on multiple cores
  - Pro: supports coverage
  - Con: not an exact representation of any WebRTC implementation
- Many bugs!

Results

- 7 vulnerabilities found and fixed
  - CVE-2018-6130 -- out-of-bounds memory issue related to in VP9
  - CVE-2018-6129 -- out-of-bounds read in VP9
  - CVE-2018-6157 -- type confusion in H264
  - CVE-2018-6156 -- overflow in FEC
  - CVE-2018-6155 -- use-after-free in VP8
  - CVE-2018-16071 -- a use-after-free in VP9
  - CVE-2018-16083 -- out-of-bounds read in FEC

CVE-2018-6130

```
std::map<int64_t, GofInfo> gof_info_ RTC_GUARDED_BY(crit_);
gof_info_.emplace(unwrapped_tl0,
    GofInfo(&scalability_structures_[current_ss_idx_],
    frame->id.picture_id));
if (frame->frame_type() == kVideoFrameKey) {
    GofInfo info =
       gof_info_.find(codec_header.tl0_pic_idx)->second;
    FrameReceivedVp9(frame->id.picture_id, &info);
    UnwrapPictureIds(frame);
    return kHandOff;
  }
```

CVE-2018-6130

```
std::map<int64_t, GofInfo> gof_info_ RTC_GUARDED_BY(crit_);
gof_info_.emplace(unwrapped_tl0,
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    FrameReceivedVp9(frame->id.picture_id, &info);
    UnwrapPictureIds(frame);
    return kHandOff;
  }
```

Google

CVE-2018-6130

const_iterator std::map::find (const key_type & __x) const [inline]

Tries to locate an element in a map.

Parameters:

x Key of (key, value) pair to be located.

Returns:

Read-only (constant) iterator pointing to sought-after element, or end() if not found.

WebRTC Security Problems

- WebRTC has billions of users
- WebRTC provided no way to report security bugs
- WebRTC documentation provided no guidance on updates

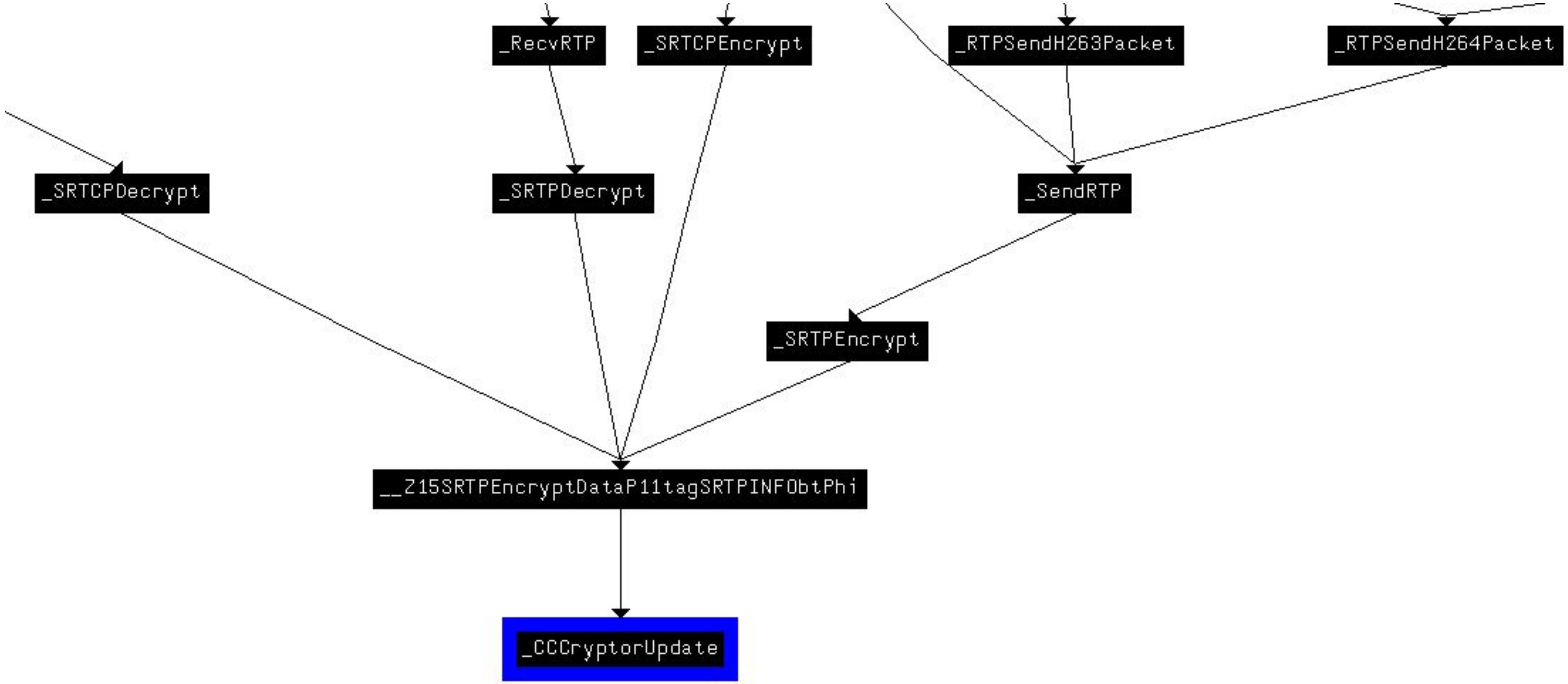
FaceTime

FaceTime

- FaceTime is closed-source and proprietary
- Needed to modify binary to log packets

FaceTime Encryption

- Used IDA to identify call to encryption function



Hooking Functions on MacOS

- CCCryptorUpdate seemed a good candidate for recording RTP
- DYLD_INTERPOSE can be used to redirect library calls on Macs
- Requires setting an environment variable
  - This isn’t possible for AVConference, which is started as a daemon

Hooking Functions on MacOS

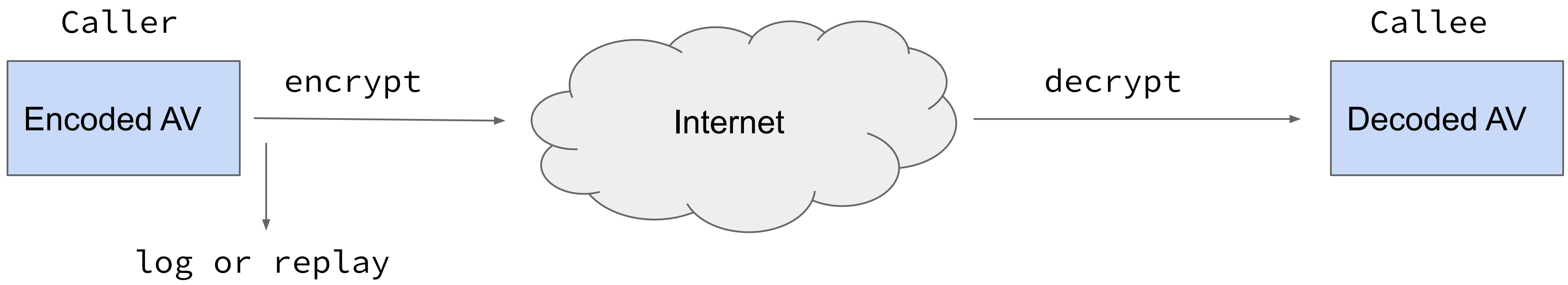
- DYLD_INTERPOSE can also be called in the static section of a library loaded by a Mac binary
- Found insert_dylib on github
  https://github.com/Tyilo/insert_dylib
- Inserted static library that hooked CCCryptorUpdate

```
DYLD_INTERPOSE(mycryptor, CCCryptorUpdate);

CCCryptorStatus mycryptor(
   CCCryptorRef cryptorRef, const void
*dataIn,
   size_t dataInLength, void *dataOut,

   size_t dataOutAvailable,size_t
*dataOutMoved) {
```

Hooking Functions on MacOS

- Tried making a call
- Needed some refinement
  - Limited hooking to functions that sent RTP
  - Added a spinlock
  - Patched binary to pass length
- Could alter RTP in real time, but replay did not work!

Hooking Functions on MacOS



Investigating RTP Packets

- Read through _SendRTP function to figure out packet generation
- Discovered RTP headers were created well after encryption

Bit Offset	0-1	2	3	4-7	8	9-15	16-31
0	Version	Padding	Ext.	CSRC Count	Marker	Payload Type	Sequence Number
32	Timestamp						
64	Synchronization Source (SSRC) Identifier						
96	Contributing Source (CSRC) Identifier						
96+32*CC	Payload						

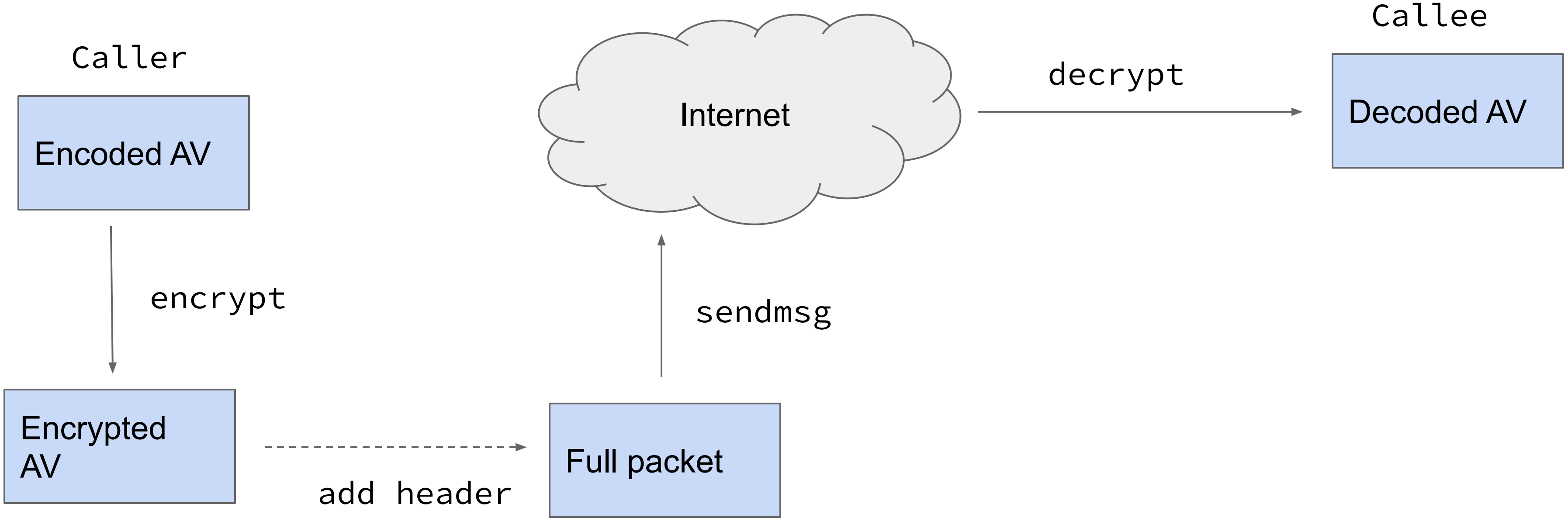
Interesting Parts of RTP Headers

- SSRC is a random identifier that identifies a stream
  - FaceTime cannot be limited to a single stream
- Payload type is a constant that identifies content type
- Extensions are extra information that is independent of the stream data
  - Screen orientation
  - Mute
  - Quality
  - Wait a sec, these totally depend on stream data

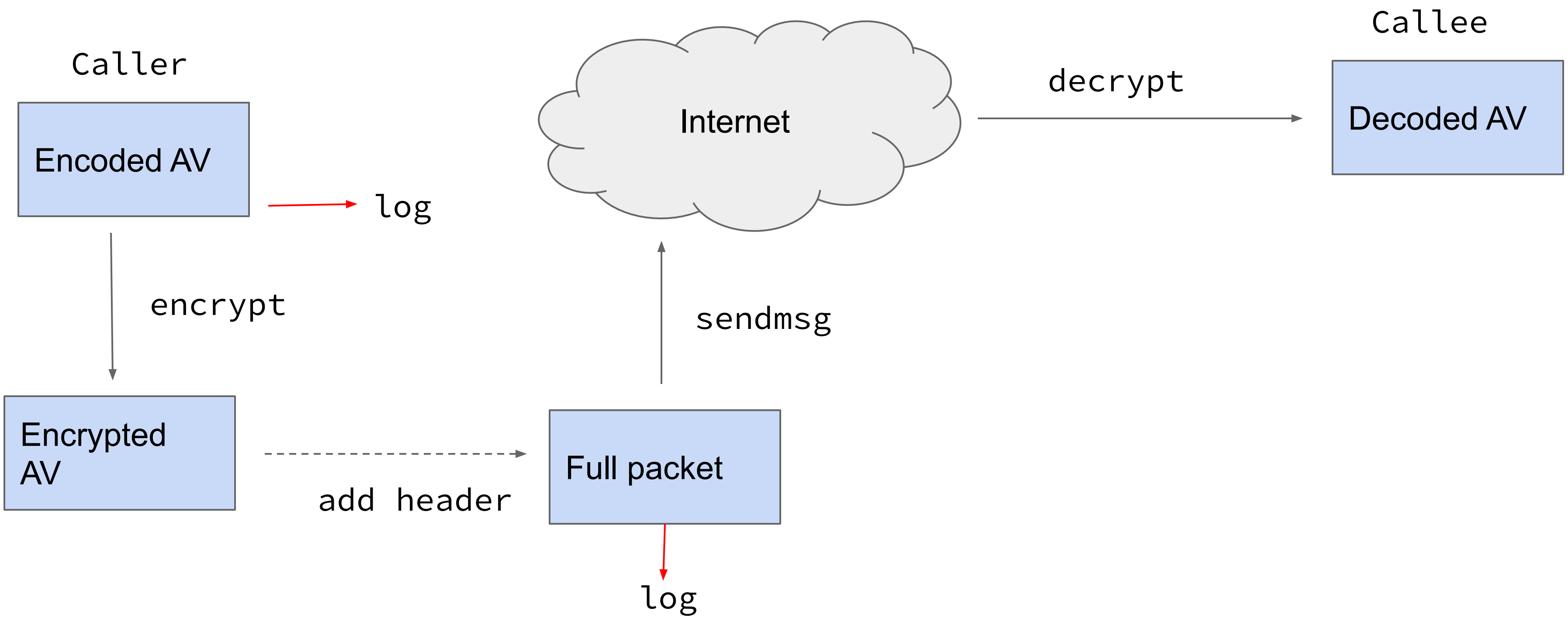
Hooking Headers?

- Tried replaying with existing headers
- Hooked sendmsg to capture and log header
  - Needed to tie encrypted message to header
  - sendmsg NOT called on packets in the same order as encryption (even with a spinlock)
  - Need to ‘fix’ SSRC and sequence number

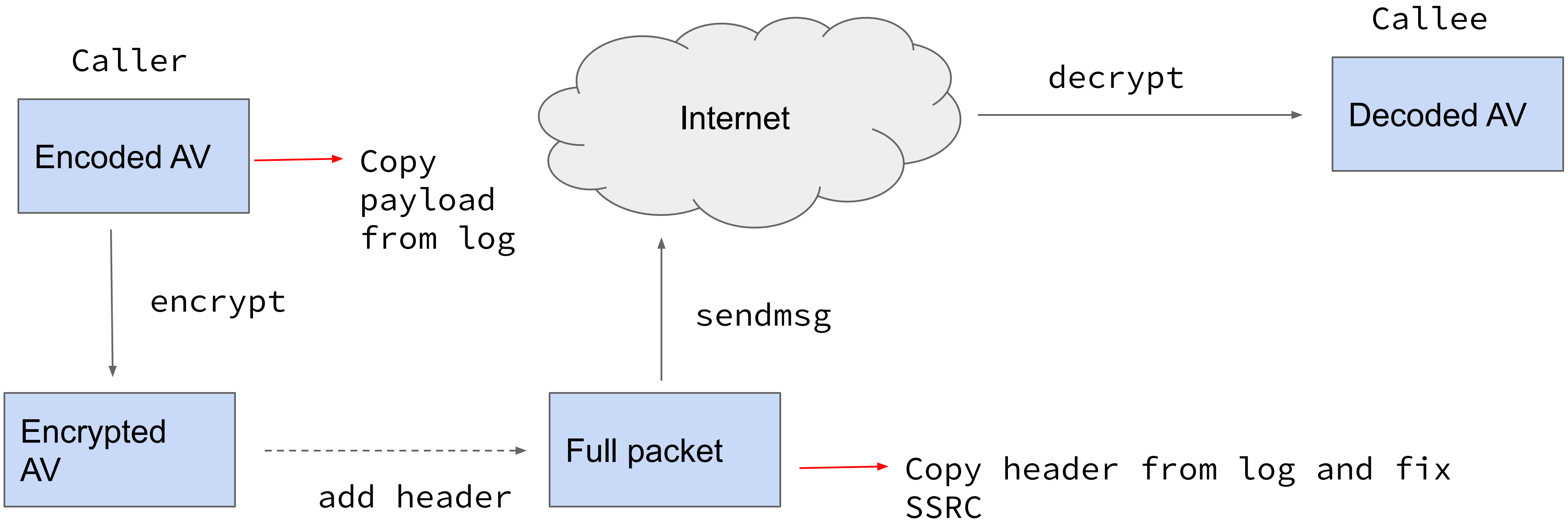
Fixing headers



Fixing headers (send)



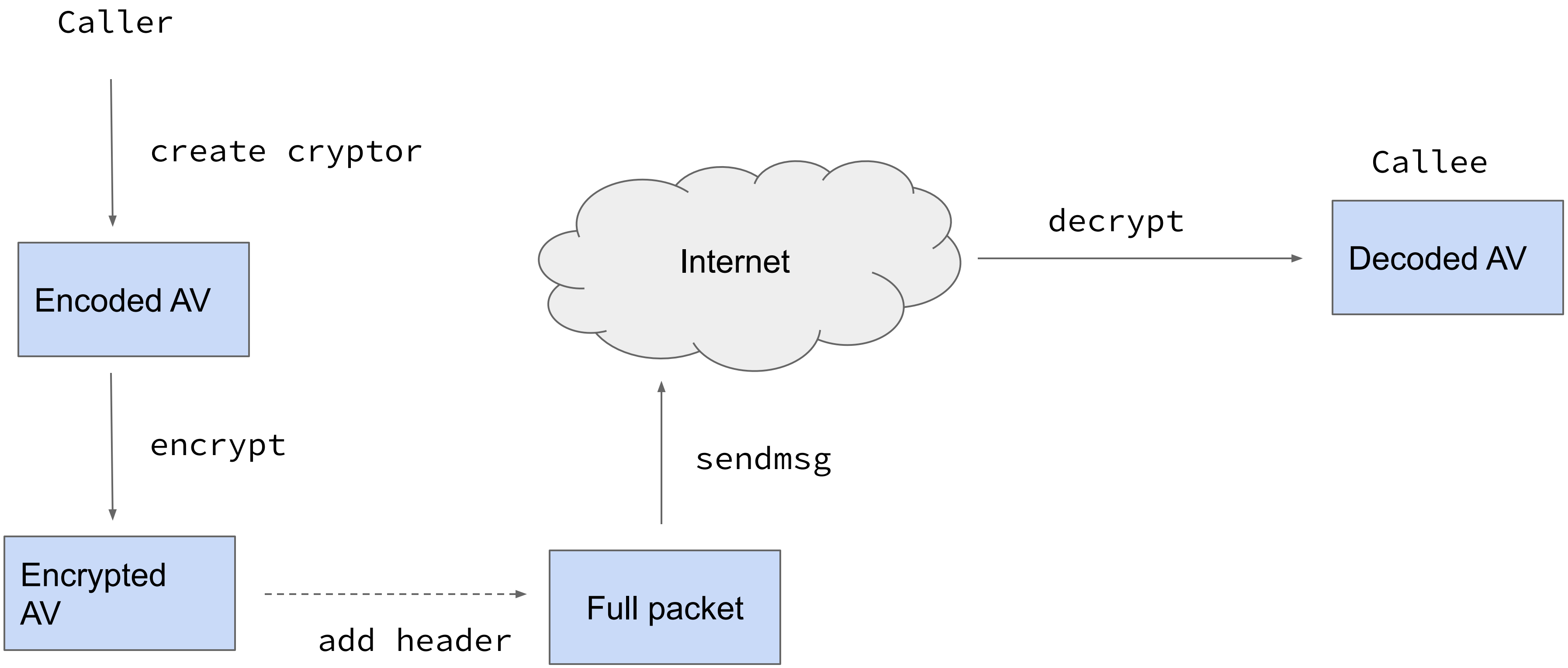
Fixing headers (replay)



Still Didn’t Work

- Patched endpoint to remove encryption
  - This worked, but can’t do it on an iPhone
  - Audio data clearly getting corrupted in decryption
- Created a cryptor queue for each SSRC, and encrypted the data in order
- Discovered encryption is XTS with sequence number as counter
- Fixed seq number counter

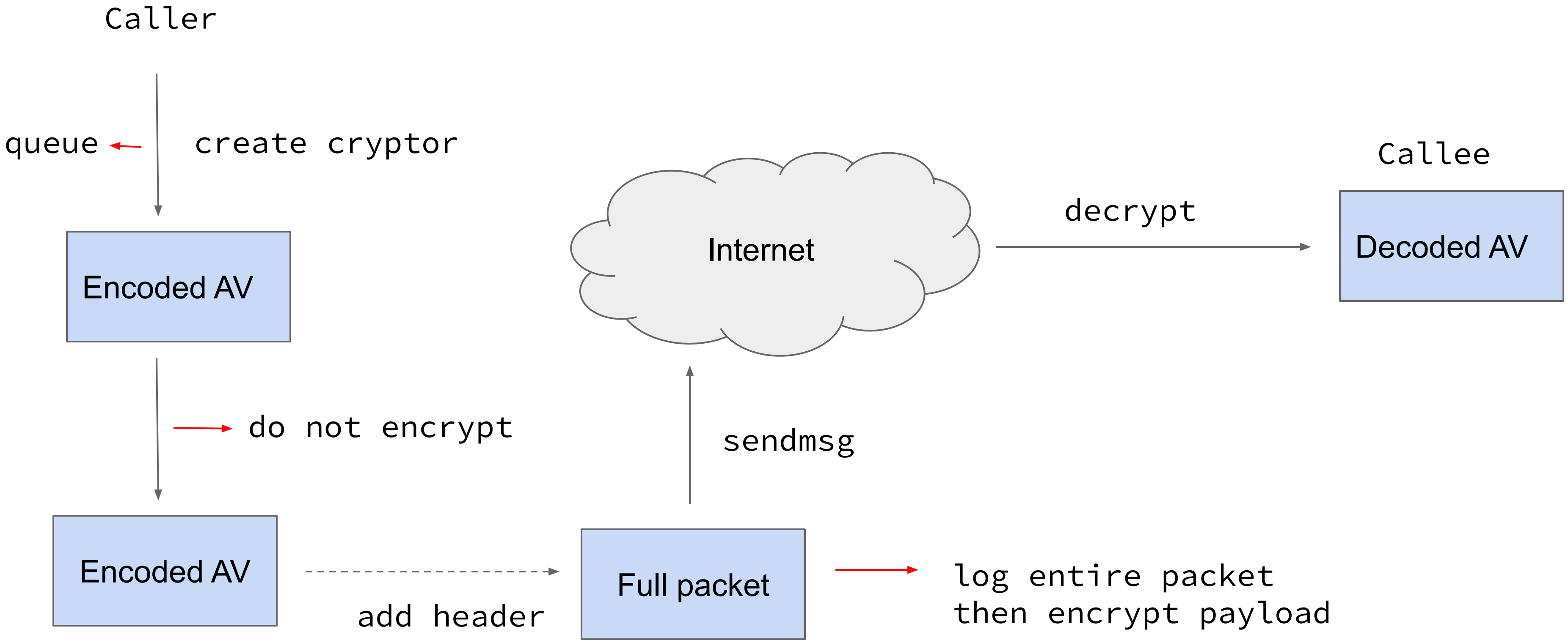
Fixing headers



Steps to Log

- Hook CCCryptorCreate to log cryptors as they are created
    - Store cryptors by thread in queues
- Hook CCCryptorUpdate, and prevent packets from being encrypted
- Hook sendmsg, log unencrypted packet, and then encrypt it using the cryptor from the queue

Fixing headers (send)



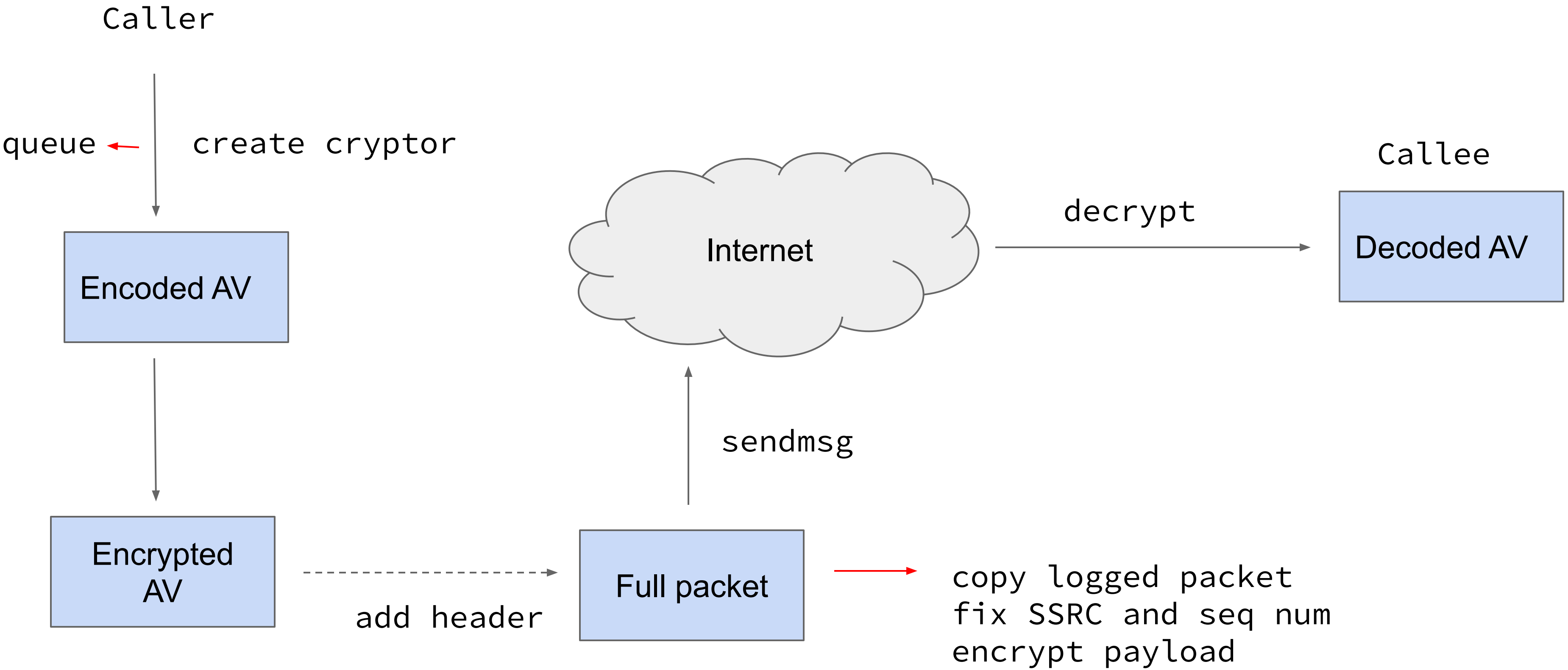
Steps to Replay

- Hook CCCryptorCreate to log cryptors as they are created
  - Store cryptors by thread in queues
- Hook sendmsg, save current ssrc and sequence number if it hasn’t been seen before
- Copy logged packet into current packet

Steps to Replay

- Replace logged ssrc with ssrc for payload type
- Replace logged sequence number with logged sequence number - starting logged sequence number + starting sequence number for ssrc
- Pop a cryptor for the payload type and encrypt the payload
    - If there are no cryptors left, don’t send and wait

Fixing headers (replay)



Demo

Results

- CVE-2018-4366 -- out-of-bounds read in video processing on Mac
- CVE-2018-4367 -- stack corruption
- CVE-2018-4384 -- kernel heap corruption in video processing
    - CVE-2015-7006 (found by Adam Donenfeld of Zimperium) is similar and exploitable
- CVE-2019-6224 -- overflow in splitting RED packets

WhatsApp

WhatsApp

- Looked at Android App
  - Desktop app does not do voice
- No symbols, but log entries from libsrtp and PJSIP
  - PJSIP is a commercial library similar to WebRTC
- Identified memcpy from packet to buffer before encryption (looked for srtp_protect log entries)

WhatsApp

- Wrote a Frida script that hooked all memcpy instances
- Frida is awesome!

```
hook_code =""

                Interceptor.attach (Module.findExportByName (
            "libc.so", "read"), {

                    onEnter: function (args) {

                    send (Memory.readUtf8String (args [1]));

                 },

                    onLeave: function (retval) {
```

WhatsApp

- Frida is too slow to make a call without a lot of lag
  - Good for debugging binary changes though
- Changed specific memcpy to point to function I wrote in ARM64
- Assembly of my function overwrote GIF transcoder

WhatsApp

- Had issues with calls disconnecting, turned out I was corrupting a used register
- After a few fixes could log and alter incoming packets
- Replaying packets by pure copying did not work

WhatsApp

- WhatsApp has FOUR RTP streams, even when muted
- Luckily, they have different payload types
- Fixing ssrc and sending logged packets worked

Crash Detection

- WhatsApp handles signal crashes internally
    - Creates crash reports in unknown format
    - FB Messenger and other apps also do this
- WhatsApp crashes do not get logged by logcat
- Stubbed out signal() and sigset() in library to get around this
- Crashes were logged by Android after this

Result

- CVE-2018-6344 -- Heap Corruption in RTP Processing

WhatsApp Signalling

- While reversing RTP processing, it became clear signalling messages were processed by native code
- Processing was not limited to correct packets for the state
- Reviewed each entry point
- Found boring crashes, but nothing interesting
    - Service respawns

WhatsApp Signalling

- Discovered signalling processes a large JSON blob "voip_params" from the server
- Sets dozens of parameters internally
- Discovered a peer could send this blob in one packet type
- Reviewed the code
- Fuzzed the parser with help from Tavis Ormandy
- No bugs ...

WhatsApp Signalling

- WhatsApp was aware of these attack surfaces
- Was aware of other voip_params issues
  - Fixed the one I reported quickly
  - Considering signing
- Has plans to reduce the attack surface of signalling

Conclusions

COMMENTARY

****, I Was Supposed To Have Learned Something From Fuzzing RTP, Wasn't I?

Scott Ippolito
11/03/15 9:51am • SEE MORE: OPINION

Scott Ippolito

Google

Bug Summary

- WebRTC: 7 bugs
- FaceTime: 5 bugs
- WhatsApp: 1 bug

Bug Location

- RTP: 0
- Error correction: 3
- Payload format: 7
- Codec: 2

Timing

- WebRTC: 4 weeks
- FaceTime: 6 weeks
- WhatsApp RTP: 2 days
- WhatsApp signalling: 3 weeks

Conclusions

- Video conferencing contained many vulnerabilities
  - Complexity is a cause, but probably necessary
  - Patching is a concern
- Video conferencing lacks test tools
  - Tooling was time consuming but worth it
  - https://github.com/googleprojectzero/Street-Party
- Signaling is a possible area for more bugs
- RTP needs more fuzzing

Questions

https://googleprojectzero.blogspot.com/

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