Communication Networks 2

SS 2021

Assignment 2

Group 06

Name	Mat.Nummer
Paul Kloker	12034928
Juan Aramis Oposich	11701238

May 27, 2021

1 Task description

The task of this assignment is to set up a Voice over IP (VoIP) Client and compare the influence of multimedia codecs on the Quality of Service (QoS) of video calls. Furthermore, the signaling messages of the Session Initiation Protocol (SIP) shall be analyzed to verify the correct behavior and to find a secret message of the registrar.

The comparison of the QoS is to be done once subjectively and once on the basis of self-selected network parameters, which are also to be visualized graphically.

2 Procedure

2.1 Linphone setup and SIP registration

Before a VoIP call can be done, one of the first thing is to do a SIP registration. SIP Endpoints are identified by their Address of Record, for example "sip:cn_06@cn2lab.cn.tuwien.at".

The endpoint is also defined by the location, which is identified by an IP address, port number and protocol. To receive calls, the caller needs to know the location of the callee. Since most SIP endpoints do not have a permanent, fixed, publicly-reachable IP address, they will need to register with a central server, or Registrar, so that they can receive incoming calls. This server accepts REGISTER requests and places the information it receives in those requests into the location service for the domain it handles. In figure 1

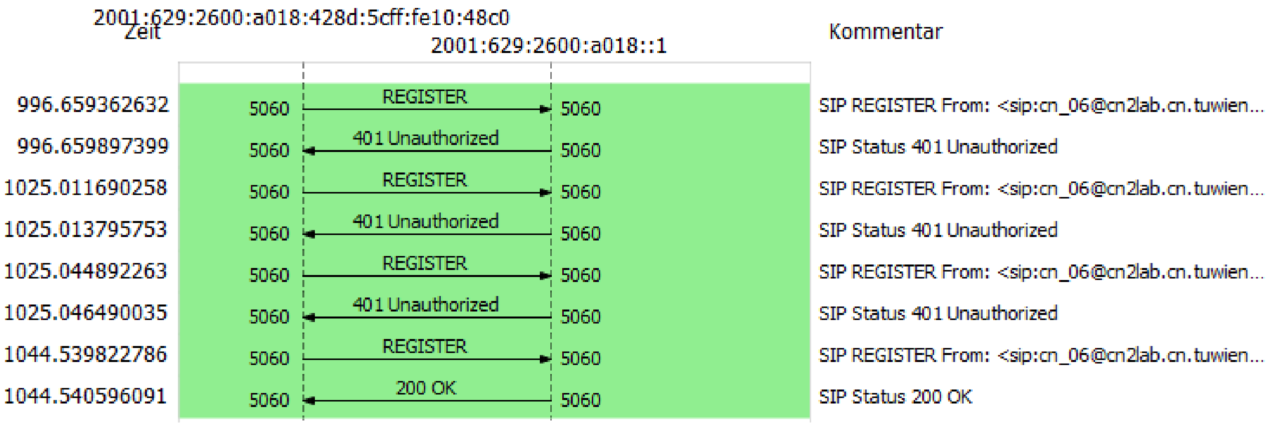

Figure 1: SIP registration flow

The first sequence is the initial registration request from SIP User Agent with it's address information. The SIP registrar answers with the information about the login "401 Unauthorized". To be authenticate, the client will combine a received nonce with his user and password information and create an MD5 hash out of them. A successful login is omitted with a 200 OK.

2.1.1 secret message

One of the task was to extract a secret message from the registration process. The secret message can be found in the last step of the registrar.

- SIP REQUEST: Digest realm="cn2lab.cn.tuwien.ac.at", nonce="YJpM2mCaS64RWAT+53QLYY40 username="cn_06", uri="sip:cn2lab.cn.tuwien.ac.at", response="aa76de94237e27cb05b0091b89a0162
- STATUS 200 OK: Secret-Message: Dazecixaru4

2.2 Capturing process

This year because of the COVID-19 pandemic it was only possible to access the lab PCs remotely. This is most of the time no problem but to compare the video and audio quality of SIP calls it is important to have direct access, because the VNC connection influences the subjective impressions. To overcome this problem a command line tool called `cn2_sbs_capture` has been provided, which captures the incoming audio and video and the outgoing video. Because the camera and microphone of the lab PC could not be used the same video was played each call.

To capture a call, the selected codecs were first set in the Linphone settings. Then `cn2_sbs_capture` was executed with the default settings and the Wireshark capture was started. After that a one-minute-long call was made. Table 1 shows all captured calls and the chosen codecs.

Because capturing all combinations of audio and video codecs would take too much time a selection of codecs was tested independent of each other. That means we assume the chosen codecs does not influence one another. This should not be done for precise performance testing.

2.3 SIP signaling and SDP

To verify that both, signaling and media works correctly and the predefined codecs are used for media transfer, a sequence diagram was created and analyzed for each capture. Figure 2 shows the sequence diagram of the SIP signaling and the media flows during the call with the VP8 video codec and the OPUS audio codec over landline. It was created by using the Wireshark SIP Flows tool. A deeper look in the SDP section in the body of the third message in the flow confirms that the offered codecs in the `INVITE` message were accepted by the callee.

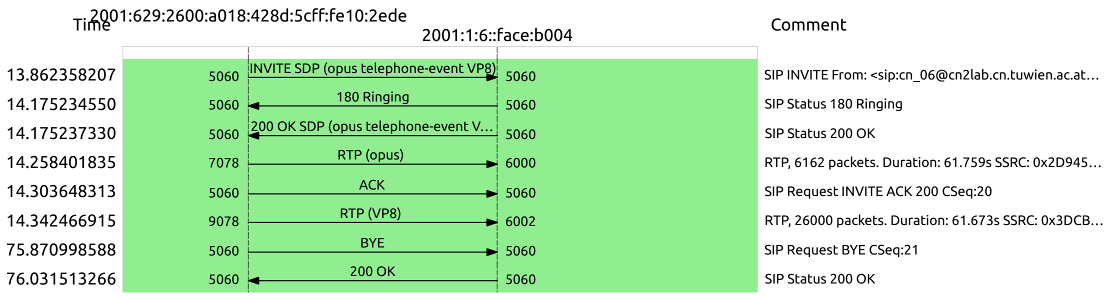

Figure 2: SIP signaling sequence diagram

In some calls, especially in the satellite calls, retransmissions of signaling messages could be identified in the auto generated sequence diagram. Another unexpected behavior which can be seen in figure 2 is the order of the `ACK` message and the initial OPUS RTP stream message. Actually, the `ACK` is sent directly after a `200 OK` of the offer and answer process but here the OPUS RTP stream starts before the acknowledgment. This behavior could be seen in most of the other captures as well, but it has no influence on the functionality because the callee already accepted the offered codecs.

Table 1: Captured calls

No.	**connection type**	**video codec**	**audio codec**
1	landline	VP8	OPUS
2	satellite	VP8	OPUS
3	landline	MP4V_ES	speex 16 kHz
4	satellite	MP4V_ES	speex 16 kHz
5	landline	MP4V_ES	PCMU
6	satellite	MP4V_ES	PCMU
7	satellite	H.263-1998_ES	GSM
8	satellite	H.263	speex 8 kHz

2.4 Audio codec quality comparison

- Audio codecs subjektiv vergleichen (vielleicht nach der Mean Opinion Score)

- RTP Analyse + Grafen

2.5 Video codec quality comparison

- Video codecs subjektiv vergleichen

z.B. `https://en.wikipedia.org/wiki/Subjective_video_quality`

Das ist so ähnlich wie MOS: https://www.irisa.fr/armor/lesmembres/Mohamed/Thesis/node146.html

- RTP Analyse + Grafen

3 Conclusion

Was sind die besten Codecs für welche Situation.