Communication Networks 2

SS 2017

Assignment 2

Group 08

Name	Mat.Nummer
Constantin SCHIEBER	01228774
Andreas HIRTENLEHNER	01327273

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1 Description of the Solution

Linphone was used and configured to register with the provided SIP Registrar (Figure 1). The provided SIP identity and password were used (`sip:cn_08@cn2lab.cn.tuwien.ac.aut`) for this (Figure 1 shows the setup dialouge).

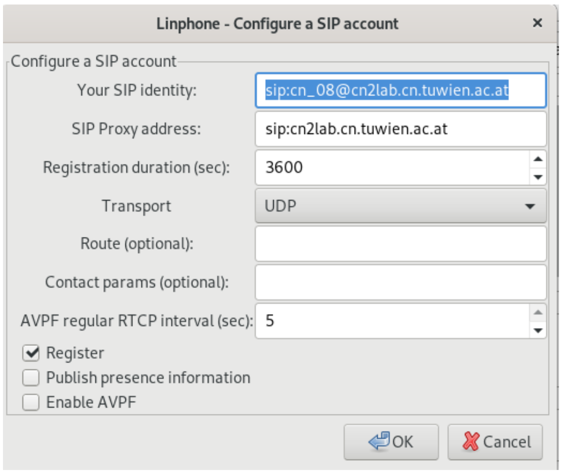

Figure 1: Setup of the Linphone client

1.1 Secret Message to the Registrar

A Wireshark log was created to detect the secret message to the registrar. A filter to `SIP` was set up to observe the communication between the Registrar and our client. We look for registration operations which enable the server to know the location of our client. For this purpose, REGISTER messages are exchanged in a regular interval and the Registrar associates our SIP identity with our currently used machine (for a detailed description refer to RFC 3261, p. 16 and the lecture slides CN2-05-SIP, p. 103). The `secret-message` field that is to be found is an unrecognised SIP header that does not conform to the standards of the protocol. It is present in the header of `200 OK` messages that are sent by the server and has the value `Dohugiwiqi3`. The secret message is therefore `Dohugiwiqi3`.

1.2 Discussion of Measured Network Parameters

Factors that influence Quality of Service (QoS) include available bandwidth, latency, packet loss, packet delay variation, out-of-order delivery and rate of corrupted packets.

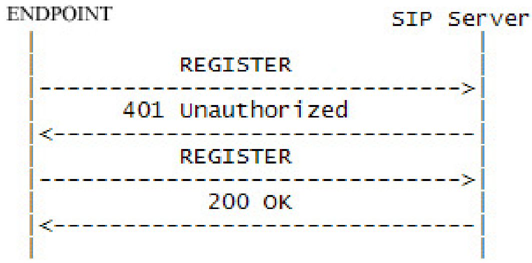

Figure 2: SIP Register Flow (https://www.voipmechanic.com/sip-call-example.htm)

Server	Payload	Packets	Lost (%)	Max Delta (ms)	Max Jitter (ms)	Mean Jitter (ms)
Landline	VP8	1553	0%	128	10.7	1.1
Landline	opus	497	0%	43	7.8	4.9
Satellite	VP8	2280	4.5%	103	7.45	3.14
Satellite	opus	589	5.3%	68	19.4	14.2

Table 1: Wireshark RTP statistics for the two servers

With Wireshark (under `Telephony > RTP > Show all streams`), we analysed lost packets, the delta between packets and the jitter (as RTP packets contain information on the senders time). Table 1 shows these statistics for both servers. They confirm our subjective observation that Landline provides a better QoS than Satellite. The call with Landline has zero lost packets while Satellite looses between 4-5% of its packets which may result in digital artefacts. A higher mean jitter for Satellite means that a bigger playback buffer is required, which results in a degradation of real-time capabilities or a stuck video stream.

1.3 Discussion of subjective QoS

We compared three different codecs for subjective video and sound quality. As expected, a changed codec for the Landline connection does not make any difference in the subjective QoS. This is because a good connection will deliver good results, even if the codec is not highly efficient / optimized (after all, the codec is designed to work as expected, at least under optimal conditions).

MP4V-ES (MPEG4, Part 2) is a codec that integrates with H.263, a standard that is optimized towards usage with a low available bandwidth (< 64kbit/s). It heavily uses temporal compression, i.e. works best when there is a low amount of movement between frames. Since H263-1998 only provides additional optional features and therefore should come out very similar to H.263.

- Landline, Opus, VP8: 5 (very good quality)
- Satellite, Opus, VP8: 3 (slight cracking in audio, video freezing every 5s)
- Satellite Speex, MP4V-ES: bad quality, fragments in picture but no freezes, lag
- Landline Speex, MP4V-ES: No difference in quality
- Satellite H263-1998, PCMU: bad quality, smoother fragments in picture but no freezes, lag
- Landline H263-1998, PCMU: No difference in quality