Telephony RTP Extractor

Project work presentation

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Introduction

- We need to open network capture file using wireshark to verify the call quality. This step is manual and need a lot of effort in manual testing the audio call quality.
- For debugging and troubleshooting it is often desirable to convert a stream of captured RTP packets to playable audio (e.g. a WAV file). A major difficulty in doing this is decoding the many different types of Codecs that can be carried in an RTP packet stream.
- **Codecs supported**: G711ulaw, G711alaw

Call Quality using MOS score

The following items can all affect call quality:

- Bandwidth
- Codec in use.
- Hardware
- Jitter
- Latency
- Packet Loss

- RTP quality is important factor to validate the SIP telephony call(audio) quality hence we need to verify the RTP Metrics like Packet Loss, Jitter, latency and come up with a predictive MoS score after **extracting the RTP payload from the network capture file.** MOS measures subjective call quality for a call. MOS scores range from 1 for unacceptable to 5 for excellent.
- VOIP calls often are in the 3.5 to 4.2 range

Analysis of SIP and RTP details

SIP Analysis

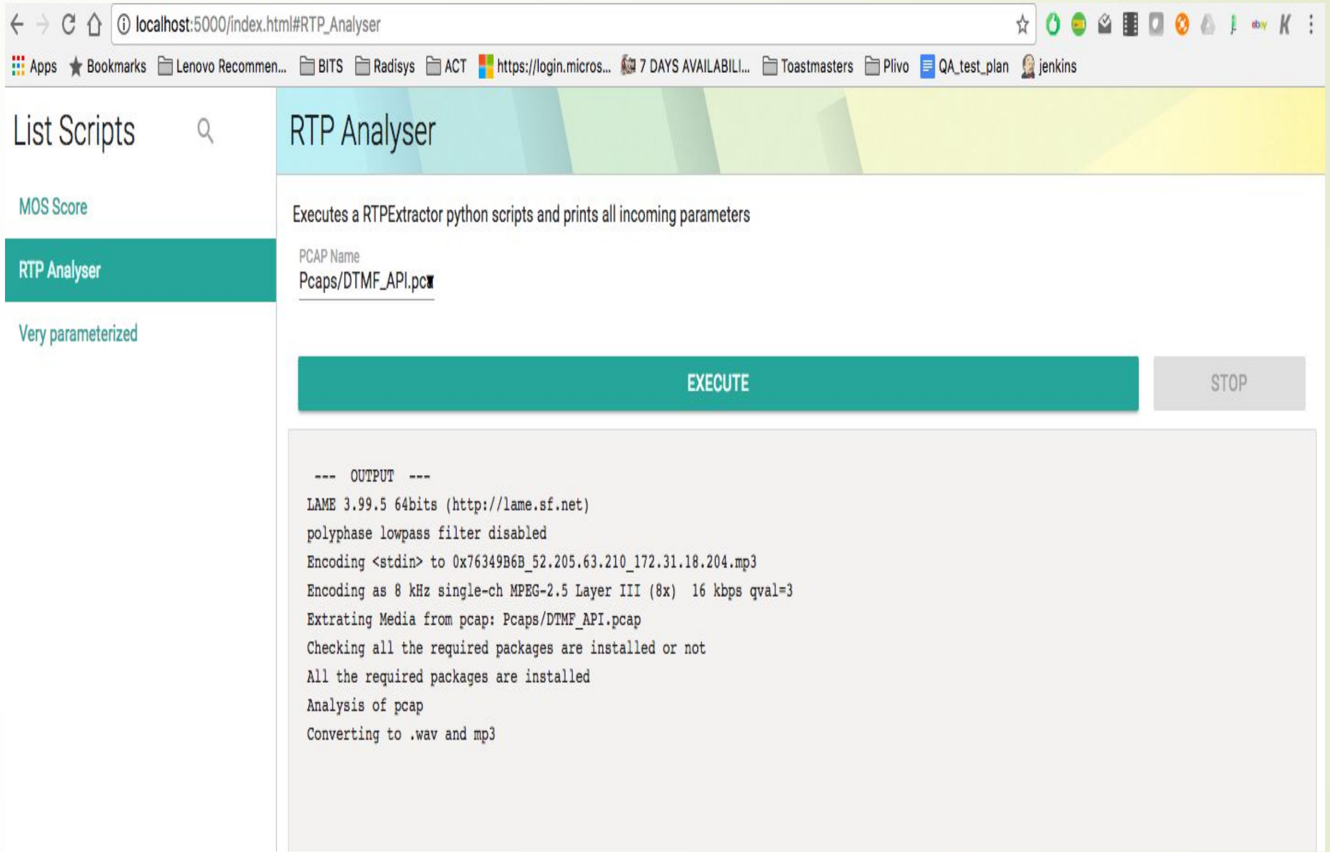
```
Chandras-MacBook-Air:call_quality_framework chandra$ cat sip_summary.txt

===================================================================
SIP Statistics

Number of SIP messages: 5
Number of resent SIP messages: 0

* SIP Status Codes in reply packets
  SIP 180 Ringing          :      1 Packets
  SIP 200 OK               :      2 Packets

* List of SIP Request methods
  ACK              :      1 Packets
  BYE              :      1 Packets

* Average setup time 0 ms
 Min 0 ms
 Max 0 ms
===================================================================
```

RTP Analysis

```
Chandras-MacBook-Air:call_quality_framework chandra$ cat rtp_summary.txt
======================== RTP Streams ========================
   Src IP addr  Port    Dest IP addr  Port       SSRC          Payload  Pkts        Lost   Max Delta(ms)  Max Jitter(ms) Mean Jitter(ms) Problems?
  52.205.63.210 24016   172.31.18.204 6000 0x76349B6B ITU-T G.711 PCMA  1146    0 (0.0%)           21.67          163.93            0.06
=============================================================
```

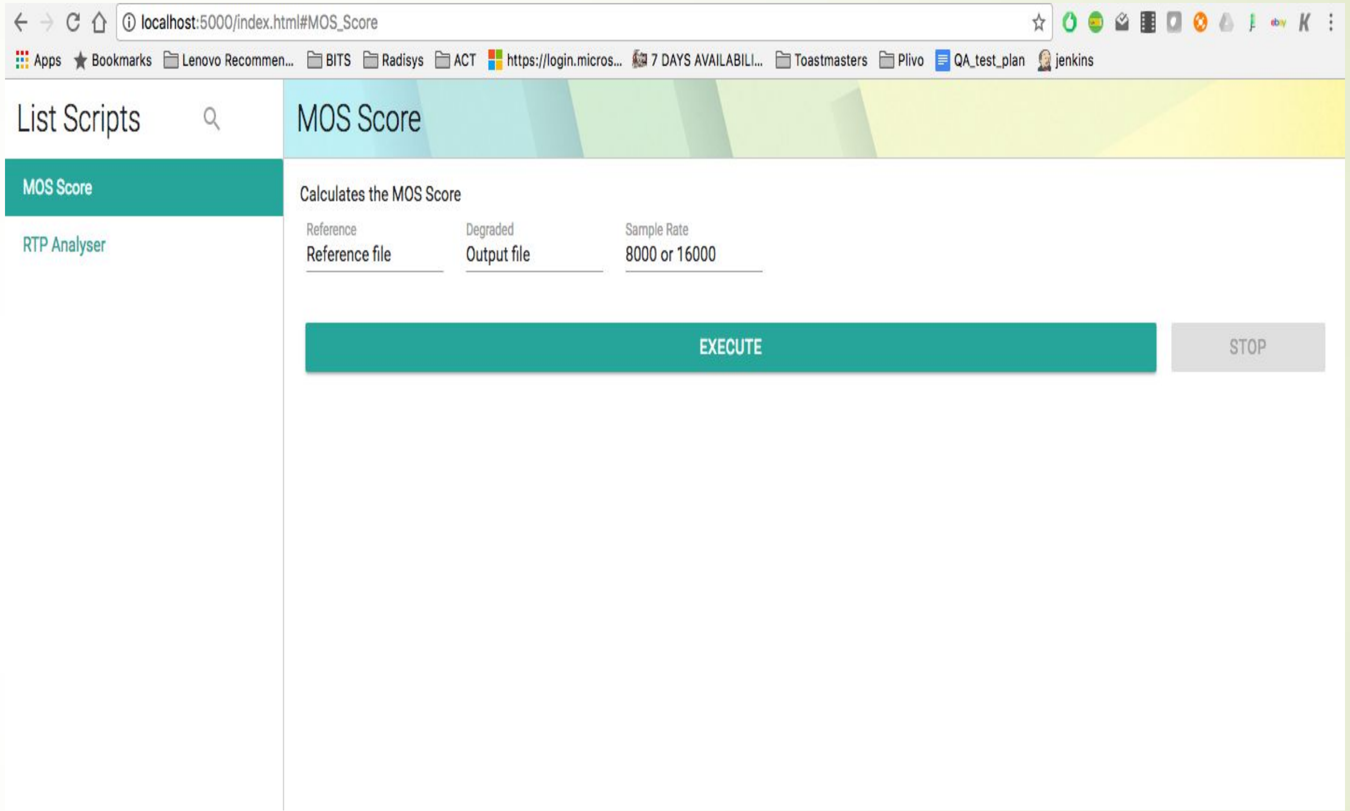
RTP Analyzer

A RTP Analyser Server provides Web GUI for your scripts and remote execution facility.We will be able to access your scripts via web-browser and execute them. Everything will run on your machine, so users shouldn't care about setting up an environment or working via ssh.



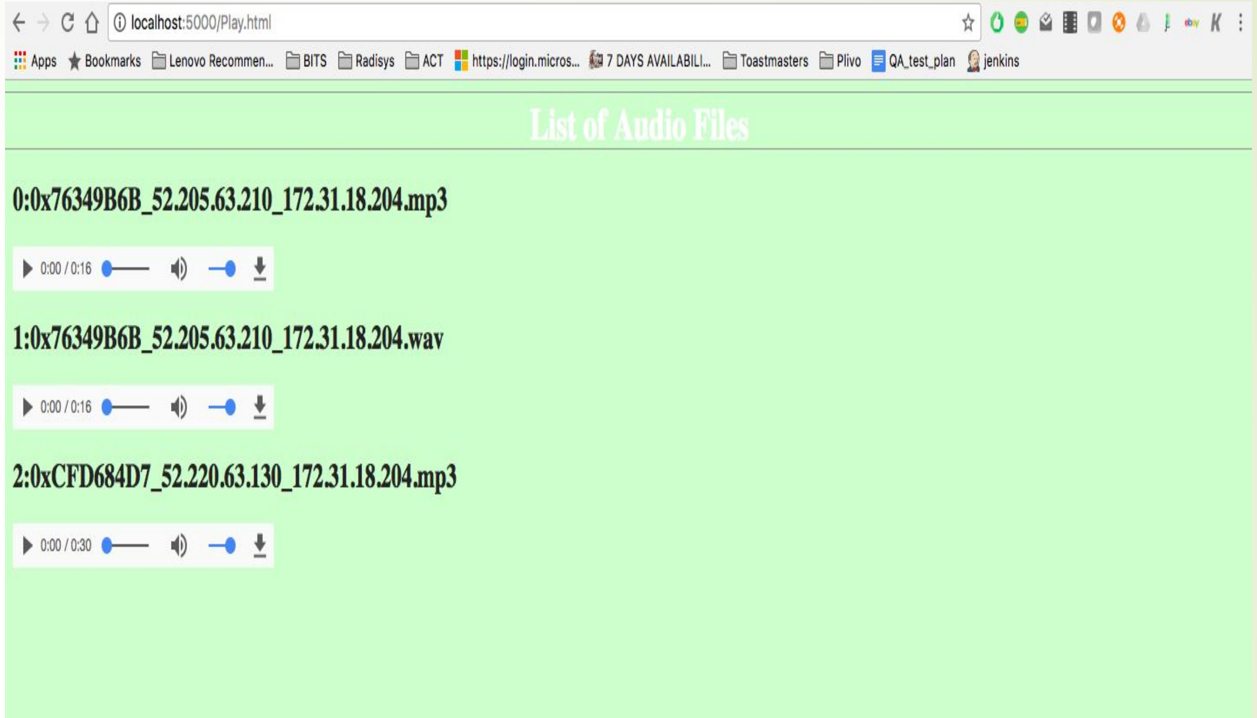
MOS Server

A MOS Server provides Web GUI for your MOS score calculation facility. We will be see MOS score in web-browser after execute it.



Audio Server

A Audio Server provides Web GUI to listen the extracted audio from server. You will be hear the audio file from the web-browser or you can download and list. Everything will run on your remote machine, so users shouldn't care about setting up an environment or working via ssh.



Direction for future work

- As of now decoding of RTP payload, works for G711 audio codec only. I will try to enhance this framework for various audio codecs like opus, amr, amr-wb evs, g722, g729 etc,

Q&A

Thank You