



Telephony RTP Extractor

Project work presentation

Presented by:-
Chandra Prakash Singh
2014HT12924



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

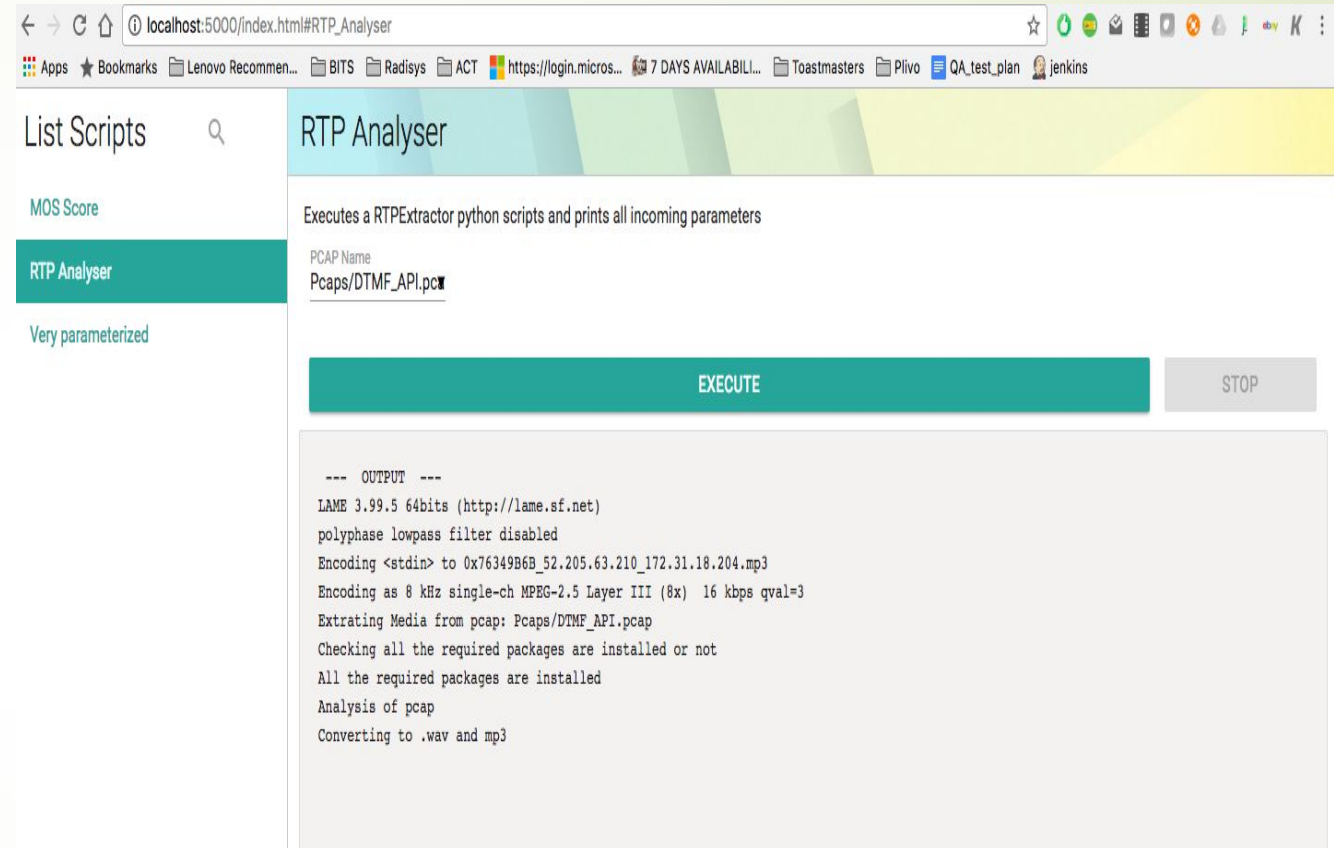
```
Chandras-MacBook-Air:call_quality_framework chandra$ cat sip_summary.txt
List of Figures
=====
SIP Statistics
Number of SIP messages: 5
Number of resent SIP messages: 0
Background
* SIP Status Codes in reply packets
  SIP 180 Ringing      : 1 Packets
  SIP 200 OK           : 2 Packets
Problem Definition
* List of SIP Request methods
  ACK                  : 1 Packets
  BYE                  : 1 Packets
Chapter 2: VoIP RTP Quality
* Average setup time 0 ms
  Min 0 ms
  Max 0 ms
  S score
=====
<proto>UDP</proto>
<src_ip>52.205.63.210</src_ip>
<dst_ip>172.31.18.204</dst_ip>
<dst_port>5060</dst_port>
<seq_no>63563</seq_no>
<ssrc>0x76349b6b</ssrc>
<proto>UDP</proto>
<src_ip>52.205.63.210</src_ip>
<src_port>24016</src_port>
<dst_ip>172.31.18.204</dst_ip>
<dst_port>6000</dst_port>
<tp>
<seq_no>63563</seq_no>
<ssrc>0x76349b6b</ssrc>
```

RTP Analysis

```
Chandras-MacBook-Air:call_quality_framework chandra$ cat rtp_summary.txt
===== RTP Streams =====
(Src IP addr Port S-M) (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 12 163.93 0.06
=====
```

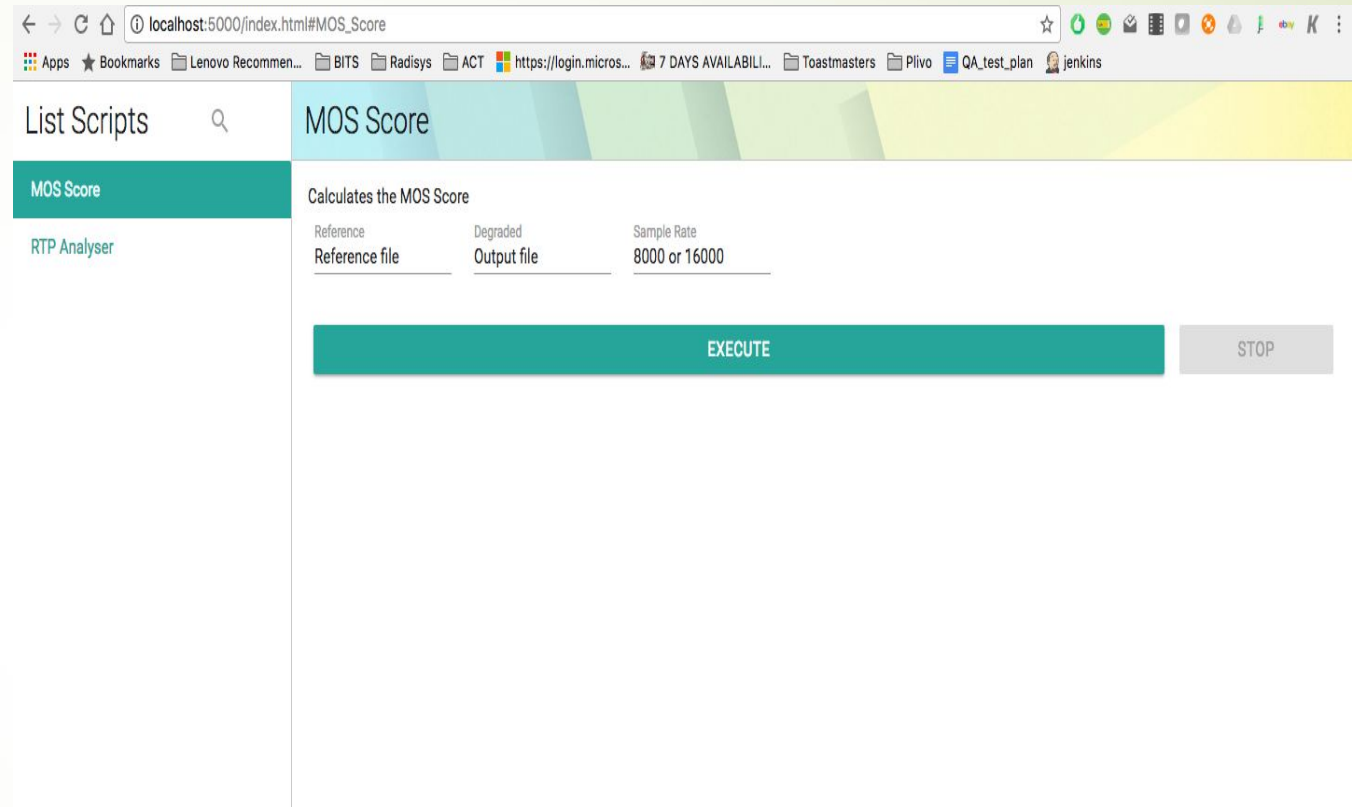
RTP Analyzer

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

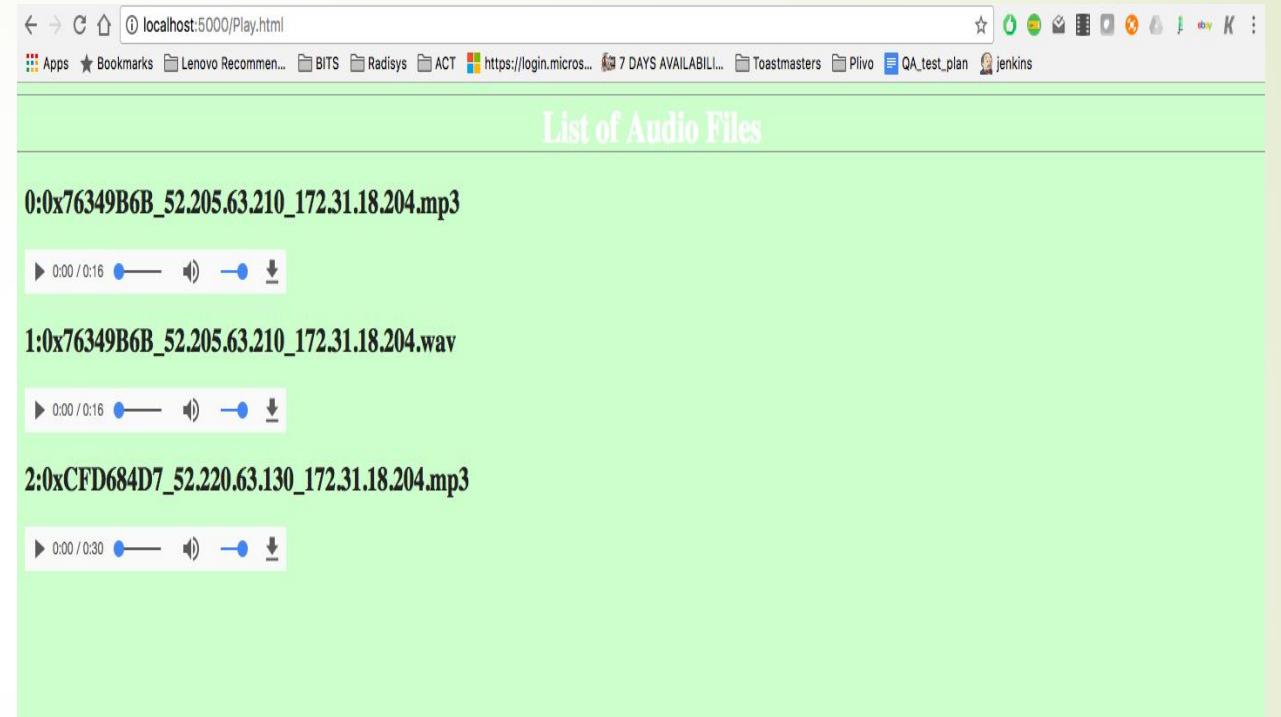
A MOS Server provides Web GUI for your MOS score calculation facility. We will be see MOS score in web-browser after execute it.



The screenshot shows a web browser window with the address bar displaying `localhost:5000/index.html#MOS_Score`. The browser's bookmark bar includes entries for 'Apps', 'Bookmarks', 'Lenovo Recommen...', 'BITS', 'Radisys', 'ACT', 'https://login.micros...', '7 DAYS AVAILABI...', 'Toastmasters', 'Plivo', 'QA_test_plan', and 'jenkins'. The web application interface has a left sidebar with 'List Scripts' and a search icon, and a main content area. The main area has a header 'MOS Score' and a sub-header 'Calculates the MOS Score'. Below this, there are three input fields: 'Reference' with a sub-label 'Reference file', 'Degraded' with a sub-label 'Output file', and 'Sample Rate' with a value of '8000 or 16000'. At the bottom of the main area, there are two buttons: a large teal 'EXECUTE' button and a smaller grey 'STOP' button. The sidebar also contains a link for 'RTP Analyser'.

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