

Sevana PCAP Analyzer - User Manual

Introduction

Sevana PCAP Analyzer (*pvqa_pcap*) is a command-line tool for analyzing VoIP call quality from network packet capture (PCAP) files. It extracts RTP audio streams, decodes them, and provides comprehensive voice quality metrics using Sevana's Perceptual Voice Quality Assessment (PVQA) technology. It tries to detect audio problems: echo, clipping, silence, background noise. The tool is capable to decode SIP signalling and extract audio streams using network & codec information from SIP.

What You Can Do With This Tool

- Analyze recorded VoIP calls for quality issues
- Get MOS (Mean Opinion Score) ratings for voice calls
- Detect audio problems: echo, clipping, silence, background noise
- Extract and decode audio streams to WAV files
- Generate detailed JSON or text reports
- Analyze SIP call flows and correlate with RTP streams

Supported Audio Codecs

- G.711 (A-law and μ -law)
 - G.729
 - iLBC
 - Opus
 - AMR / AMR-WB (may require additional licensing)
 - GSM-EFR (may require additional licensing)
 - EVS (may require additional licensing)
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Getting Started

Prerequisites

1. **License file:** You need a valid *pvqa_pcap.lic* license file from Sevana (or API key to use with the license server)
2. **Configuration file:** A *pvqa_pcap.cfg* configuration file
3. **PCAP file:** A network capture file containing VoIP traffic

Quick Start

1. Place your license file and configuration file in the same directory as the tool
2. Run the analyzer:

```
./pvqa_pcap --input your_capture.pcap --config pvqa_pcap.cfg
```

The `--config` option may be missed if configuration file is named *pvqa_pcap.cfg*

3. View the output report
-

Configuration File

The configuration file uses YAML format. Create a file named *pvqa_pcap.cfg* with the following structure:

```
pvqa:  
  license: pvqa_pcap.lic
```

License Options

The *license* field supports three formats:

Format	Example	Description
File path	<i>pvqa_pcap.lic</i>	Path to local license file
License URL	<i>license:// apikey@license.server.com</i>	License server URL

Command-Line Reference

Basic Syntax

```
pvqa_pcap [OPTIONS] --input <pcap_file> --config <config_file>
```

Required Options

Option	Description
<i>--input</i> <i><file></i>	Path to the PCAP file to analyze
<i>--config</i> <i><file></i>	Path to the configuration file

Output Options

Option	Description
<i>--json-format</i>	Generate JSON output instead of plain text
<i>--united-report</i> <i><path></i>	Save report to specified file; use <i>-</i> or <i>console</i> for stdout
<i>--audio-output</i> <i><dir></i>	Save decoded audio streams as WAV files to this directory
<i>--save-sip</i> <i><file></i>	Save extracted SIP messages to file
<i>--log</i> <i><file></i>	Write processing log to file

Processing Options

Option	Description
<i>--threads</i> <i><n></i>	Number of processing threads (default: 1). Use <i>0</i> for all CPUs
<i>--single-pass</i>	Use single-pass analysis mode (faster, uses more memory)
<i>--no-decode</i>	Skip audio decoding; generate network statistics only
<i>--verbose</i>	Show detailed processing output

Progress Display

Option	Description
<i>--progress none</i>	No progress output (default)
<i>--progress print</i>	Print progress percentage to console line by line
<i>--progress show</i>	One-line progress display

Filtering Options

Option	Description
<i>--filter-call-id <id></i>	Process only streams matching this SIP Call-ID
<i>--filter-ptype <type></i>	Process only RTP streams with this payload type (can specify multiple)
<i>--max-stream-count <n></i>	Limit analysis to first N streams
<i>--match-stream-id <id></i>	Process only the specified stream ID

SIP Options

Option	Description
<i>--grep-sip</i>	Look for SIP traffic (UDP/TCP) and correlate with RTP streams
<i>--require-sip <code></i>	Only process RTP streams that have SIP handshake with specified response code

Advanced Options

Option	Description
<i>--esp-icv-size <size></i>	ESP ICV hash size for encrypted traffic
<i>--silent-tail <seconds></i>	Check for silence at end of stream (specify duration in seconds)

Option	Description
<code>--version</code>	Display version information and exit

Usage Examples

Basic Analysis

Analyze a PCAP file and display results on console:

```
./pvqa_pcap --input call_recording.pcap --config pvqa_pcap.cfg
```

Generate JSON Report

Create a machine-readable JSON report:

```
./pvqa_pcap --input call_recording.pcap --config pvqa_pcap.cfg \
  --json-format --united-report analysis_report.json
```

Extract Audio Files

Decode and save audio streams as WAV files:

```
./pvqa_pcap --input call_recording.pcap --config pvqa_pcap.cfg \
  --audio-output ./decoded_audio/
```

High-Performance Analysis

Use all available CPUs for faster processing:

```
./pvqa_pcap --input large_capture.pcap --config pvqa_pcap.cfg \
  --threads 0 --progress show
```

Analyze Specific Call

Filter analysis to a specific SIP call:

```
./pvqa_pcap --input multi_call.pcap --config pvqa_pcap.cfg \
  --filter-call-id "abc123@192.168.1.100"
```

Network Statistics Only

Get network metrics without audio decoding (faster):

```
./pvqa_pcap --input call_recording.pcap --config pvqa_pcap.cfg \  
--no-decode --json-format
```

Full Analysis with All Outputs

Complete analysis with audio extraction, SIP logs, and JSON report:

```
./pvqa_pcap --input call_recording.pcap --config pvqa_pcap.cfg \  
--json-format \  
--united-report report.json \  
--audio-output ./audio/ \  
--save-sip sip_messages.txt \  
--log analysis.log \  
--threads 4 \  
--progress show
```

Understanding the Output Report

The JSON report is organized into several sections. Let's examine each one.

Packet Counter

The first section provides statistics about packets found in the PCAP file:

```
{  
  "packet_counter": {  
    "data": [  
      {  
        "name": "IP packets",  
        "value": "21442",  
        "info": "Number of found IP packets"  
      },  
      {  
        "name": "UDP packets",
```

```
    "value": "18889",
    "info": "Number of found UDP packets"
  },
  {
    "name": "RTP packets",
    "value": "18817",
    "info": "Number of found RTP packets (without RTCP)"
  },
  {
    "name": "RTP packets (unknown payload)",
    "value": "0",
    "info": "Number of found RTP packets without known payload type i.e.
failed to associate with voice codec"
  },
  {
    "name": "RTP packets (without SIP info)",
    "value": "0",
    "info": "Number of found RTP packets which do not belong to any SIP
call"
  },
  {
    "name": "RTCP packets",
    "value": "72",
    "info": "Number of found RTCP packets."
  },
  {
    "name": "RTP endpoints",
    "value": "7",
    "info": "Number of found RTP endpoints/tuples"
  },
  {
    "name": "SIP packets",
    "value": "123",
    "info": "Number of found SIP packets (UDP and TCP)"
  },
  {
    "name": "SIP TCP packets",
    "value": "123",
    "info": "Number of found SIP TCP packets"
  },
  {
    "name": "SIP UDP packets",
```

```
    "value": "0",
    "info": "Number of found SIP UDP packets"
  },
  {
    "name": "SIP INVITE requests",
    "value": "9",
    "info": "Number of SIP INVITE request messages"
  },
  {
    "name": "SIP INVITE responses",
    "value": "21",
    "info": "Number of SIP INVITE response messages"
  },
  {
    "name": "BYE requests",
    "value": "3",
    "info": "Number of SIP BYE request messages"
  },
  {
    "name": "BYE responses",
    "value": "0",
    "info": "Number of SIP BYE response messages"
  },
  {
    "name": "IP packets fragmented",
    "value": "0",
    "info": "Number of fragmented IP packets"
  },
  {
    "name": "IP packets non-fragmented",
    "value": "40330",
    "info": "Number of non-fragmented IP packets"
  }
]
}
```

Each entry contains *name*, *value*, and *info* fields. The *info* field provides a human-readable description of each counter.

SIP Statistics

The next section summarizes SIP call states:

```
{
  "sip_statistics": {
    "data": [
      {
        "name": "Semistarted calls",
        "value": "0",
        "info": "Calls with initial INVITE but without 200 OK"
      },
      {
        "name": "Started calls",
        "value": "3",
        "info": "Calls with initial INVITE and 200 OK response"
      },
      {
        "name": "Semicomplete calls",
        "value": "0",
        "info": "Calls with initial INVITE + 200 OK + initial BYE request"
      },
      {
        "name": "Complete calls",
        "value": "3",
        "info": "Calls with INVITE/200 OK and BYE/ACK"
      },
      {
        "name": "Finished calls",
        "value": "0",
        "info": "Calls without INVITE but with BYE"
      },
      {
        "name": "Maximal number of used RTP tuples",
        "value": "18",
        "info": "Maximal number of simultaneously used RTP peers"
      },
      {
        "name": "Bad SIP messages",
        "value": "0",
        "info": "Number of SIP messages failed to parse"
      }
    ]
  }
}
```

```
{
  "name": "Calls with media",
  "value": "3",
  "info": "Number of SIP calls with detected media RTP streams."
}
]
```

This helps you understand call completion rates and identify incomplete or failed calls in the capture.

Global Report Fields

Several top-level fields provide general information:

```
{
  "silent_tail_length_seconds": 0,
  "analysis_duration_ms": 9610,
  "software": "vq_pcap v1.8.10, build 10991"
}
```

Field	Description
<i>silent_tail_length_seconds</i>	Configured silent tail detection length (from <i>--silent-tail</i> option)
<i>analysis_duration_ms</i>	Total processing time in milliseconds
<i>software</i>	Version of the analyzer that generated this report

Streams Array

The *streams* array contains detailed information for each RTP stream. Here's an example stream:

```
{
  "streams": [
    {
      "stream_id": 1,
      "rtp_ssrc": 575474971,
      "rtp_src": "192.168.2.93:26484",
```

```
"rtp_dest": "192.168.2.93:18416",
"mos_sevana": 3.84458,
"mos_network": 4.14,
"rtp_counter": 3135,
"rtp_lost": 0,
"rtp_illegal": 0,
"rtcp_counter": 12,
"jitter": 0.0416667,
"delay": 0.152,
"codec": "OPUS",
"starttime": 1763112574298,
"endtime": 1763112636998,
"is_silent": false,
"audio": "audio/s_00000001.wav",
"sip_callid": "fSHCB9emgQqgFDqnwgrVgA..",
"sip_peer_1": "alice@sip.sevana.biz:5060",
"sip_peer_2": "bob@sip.sevana.biz",
"sip_codec": "OPUS ptype: 107, rate: 48000, channels: 2\n",
"sip_bye": true,
"ptypes": "ptype: 107 : 3135 ",
"detector_report": { ... }
}
]
```

Stream identification:

Field	Description
<i>stream_id</i>	Unique identifier for this stream
<i>rtp_ssrc</i>	RTP Synchronization Source identifier
<i>rtp_src</i>	Source IP address and port
<i>rtp_dest</i>	Destination IP address and port

Quality scores:

Field	Description
<i>mos_sevana</i>	Sevana MOS score (1.0-5.0) - perceptual audio quality

Field	Description
<i>mos_network</i>	Network MOS score (1.0-5.0) - based on network metrics

Network statistics:

Field	Description
<i>rtp_counter</i>	Total number of RTP packets in stream
<i>rtp_lost</i>	Number of lost packets
<i>rtp_illegal</i>	Number of packets with sequence/timing issues
<i>rtcp_counter</i>	Number of RTCP packets
<i>jitter</i>	Packet jitter in seconds
<i>delay</i>	Network delay in seconds

Stream details:

Field	Description
<i>codec</i>	Detected audio codec (e.g., OPUS, G711A, G729)
<i>starttime</i>	Stream start time (Unix timestamp in milliseconds)
<i>endtime</i>	Stream end time (Unix timestamp in milliseconds)
<i>is_silent</i>	<i>true</i> if stream contains only silence
<i>audio</i>	Path to decoded WAV file (if <i>--audio-output</i> used)
<i>ptypes</i>	RTP payload types found in stream with packet counts

SIP correlation:

Field	Description
<i>sip_call_id</i>	Associated SIP Call-ID

Field	Description
<i>sip_peer_1</i>	First SIP endpoint (caller)
<i>sip_peer_2</i>	Second SIP endpoint (callee)
<i>sip_codec</i>	Codec information from SIP negotiation
<i>sip_bye</i>	<i>true</i> if call was properly terminated with BYE

Quality Scores (MOS)

MOS (Mean Opinion Score) ranges from 1.0 to 5.0:

Score	Quality	Description
4.3 - 5.0	Excellent	Imperceptible degradation
4.0 - 4.3	Good	Perceptible but not annoying
3.6 - 4.0	Fair	Slightly annoying
3.1 - 3.6	Poor	Annoying
2.6 - 3.1	Bad	Very annoying
1.0 - 2.6	Very Bad	Unacceptable

Sevana MOS (*mos_sevana*) reflects actual perceived audio quality using advanced signal analysis.

Network MOS (*mos_network*) estimates quality based on network conditions (packet loss, jitter).

Detector Report

Each stream includes a *detector_report* with detailed quality analysis:

```
{
  "detector_report": {
    "mos": 3.84458,
    "rfactor": 0.923913,
    "imps": [
      {
        "Echo": 7
      }
    ],
    "echo": [
      {
        "time_ms": 12490,
        "level": -25.8475,
        "length": 1.86663
      },
      {
        "time_ms": 12990,
        "level": -25.8475,
        "length": 1.86663
      },
      {
        "time_ms": 13850,
        "level": -27.8321,
        "length": 2.048
      }
    ],
    "headers": [ ... ],
    "rows": [ ... ]
  }
}
```

Field	Description
<i>mos</i>	Overall MOS score for this stream
<i>rfactor</i>	R-Factor (transmission rating factor, 0-1 scale)
<i>r</i>	
<i>imps</i>	Array of detected impairments with occurrence counts

Field	Description
<i>echo</i>	Detailed echo events (see below)

Echo events provide precise timing and characteristics of detected echo:

Field	Description
<i>time_ms</i>	Time offset in milliseconds where echo was detected
<i>level</i>	Echo level in dB (more negative = quieter echo)
<i>length</i>	Echo tail length in seconds

Time-Segmented Analysis

The *headers* and *rows* arrays provide time-segmented quality analysis:

```
{
  "headers": [
    "Time",
    "SNR",
    "DeadAir-00",
    "DeadAir-01",
    "Click",
    "VAD-Clipping",
    "AmpClipping",
    "DynClipping",
    "Echo",
    "SilentCall",
    "Status"
  ],
  "rows": [
    [
      "0.000 : 0.680",
      "0.000",
      "0.000",
      "0.000",
      "0.000",
      "0.000",
      "0.000",
      "0.000",
      "0.000"
    ]
  ]
}
```

```

    "0.000",
    "1.000",
    "Uncertain"
  ],
  [
    "1.360 : 2.040",
    "0.000",
    "0.000",
    "0.000",
    "0.000",
    "0.000",
    "0.000",
    "0.000",
    "0.000",
    "0.000",
    "0.478",
    "Ok"
  ]
]
}

```

Each row represents approximately 680ms of audio. The columns are:

Column	Description
<i>Time</i>	Time segment (start : end in seconds)
<i>SNR</i>	Signal-to-Noise Ratio issues (0.0 = good, 1.0 = severe)
<i>DeadAir-00</i>	Complete silence in both directions
<i>DeadAir-01</i>	Silence where speech was expected (one direction)
<i>Click</i>	Click/pop artifacts detected
<i>VAD-Clipping</i>	Voice Activity Detection cutting off speech
<i>AmpClipping</i>	Amplitude clipping from overdriven signal
<i>g</i>	
<i>DynClipping</i>	Dynamic range compression artifacts
<i>g</i>	
<i>Echo</i>	Echo detection level
<i>SilentCall</i>	Silent call detection (1.0 = silence, lower = speech present)
<i>Status</i>	Segment quality: <i>Ok</i> , <i>Uncertain</i> , or <i>Poor</i>

Detector values range from 0.0 (not detected) to 1.0 (severe). Use this data to identify exactly when and where quality issues occurred during the call

Processing Modes

Multi-Pass Mode (Default)

The default mode processes PCAP files in three passes:

1. **Pass 1:** Scans the file to identify all streams and SIP calls
2. **Pass 2:** Decodes audio and performs quality analysis
3. **Pass 3:** Correlates SIP calls with RTP streams and generates report

This mode is memory-efficient for large files.

Single-Pass Mode

Use `--single-pass` for:

- Live capture analysis
 - When memory & CPU are not the constraints
- ```
./pvqa_pcap --input capture.pcap --config pvqa_pcap.cfg --single-pass
```
- 

## Troubleshooting

### Common Issues

#### “Config file not found”

- Ensure `pvqa_pcap.cfg` exists in the specified path
- Check file permissions

#### “License error” or analysis fails silently

- Verify license file path in configuration
- Check license file is valid and not expired
- For license server: verify network connectivity

### No streams detected

- Ensure PCAP contains RTP traffic
- Try with `--grep-sip` to help identify SIP-correlated streams
- Check if traffic uses supported codecs

### Low MOS scores on known-good calls

- Verify the correct codec is being detected
- Check for packet loss in the capture itself
- Ensure complete call was captured

### High memory usage

- Use default multi-pass mode (avoid `--single-pass`)
- Limit streams with `--max-stream-count`
- Process smaller time segments

### Slow processing

- Increase threads with `--threads 0` (use all CPUs)
- Use `--no-decode` if only network statistics needed
- Filter to specific calls with `--filter-call-id`

## Getting Help

For additional support:

- Check the included PDF documentation in the *docs/* folder
  - Contact Sevana support with your license information
- 

*Sevana PCAP Analyzer - Voice Quality Assessment Tool*