 # **Report on RTP Payload Discrepancy Leading to Call Silence

1. Overview

This report examines an issue involving Real-time Transport Protocol (RTP) streams exchanged between the source IP **192.10.1.65** and the destination IP **208.93.9.179**. The primary issue identified is the transmission of an oversized payload, which deviates from the expected standard, causing periods of silence during the call.

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2. Issue Summary

The client was expected to send an RTP payload with a size of **20 ms**. However, the captured data shows that the payload size transmitted was **30 ms**. This inconsistency directly contributed to silence in the call, as the Vega system was unable to properly process the larger-than-expected payload.

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3. Call Flow Analysis

Below is a summary of the key events from the SIP and RTP flows:

- **SIP INVITE** from **192.10.1.65** to **208.93.9.179** initiated the call.
- The session was established successfully with a **200 OK** response, and RTP streams began flowing between the two endpoints.
- RTP streams with **SSRC 0x19A85265** and **SSRC 0x4D3C8D43** were exchanged between **192.10.1.65:10044** and **208.93.9.179:24342**.

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4. RTP Analysis

A review of the RTP packets revealed the following key details:

Source Address	Source Port	Destination Address	Destination Port	SSRC	Start Time	Duration (s)	Payload	Packets	Lost	Min Delta (ms)	Mean Delta (ms)	Max Delta (ms)	Mir Jitt
192.10.1.65	10044	208.93.9.179	24342	430461541	55.15717	38.651138	g711U	1285	2	29.776	30.102	100.001	0.0(
192.10.1.65	10040	208.93.9.179	42138	3331846846	4.19609	32.870571	g711U	1094	1	29.224	30.074	110.043	0.0:
208.93.9.179	42138	192.10.1.65	10040	3985705944	4.18013	32.920705	g711U	1645	2	16.118	20.025	40.106	0.0(
208.93.9.179	24342	192.10.1.65	10044	1295813955	55.21338	38.659701	g711U	1933	1	18.299	20.010	40.024	0.0(

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5. Root Cause Analysis

The root cause of the problem is the deviation in the RTP payload size. The client was expected to transmit a **20 ms** payload but instead transmitted a **30 ms** payload. The Vega system, which is configured to expect a **20 ms** payload, could not handle the larger payload size. This caused momentary silences during playback as Vega had to buffer or drop excess data.

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6. Technical Explanation

RTP payload size impacts the timing of audio packetization. When the payload size exceeds the expected duration, the receiving system's jitter buffer may overflow or be unable to decode the payload properly. This misalignment results in call silences or audio clipping. The following discrepancies were observed:

- **SSRC 0x19A85265**: Expected **20 ms**, but received **30 ms**, impacting **1285 packets**.
- **SSRC 0x4D3C8D43**: Same discrepancy of **30 ms** vs. **20 ms**, affecting **1933 packets**.

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7. Impact Analysis

The increased payload size affected the RTP streams' consistency and call quality. End users likely experienced silence or choppy audio during the call. The issue was further exacerbated by packet loss (up to **0.16%**) on certain streams, compounding the impact on call quality.

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8. Recommendations

To prevent similar issues in the future, the following recommendations are made:

1. **Payload Configuration**: Ensure that the client's RTP payload size is correctly set to **20 ms**.
2. **System Alerts**: Implement monitoring alerts to detect when RTP payload size exceeds the configured threshold.
3. **Vega Configuration**: Verify that the Vega system's jitter buffer can accommodate slight deviations in payload size, if possible.
4. **Testing and QA**: Perform end-to-end call testing after configuration changes to verify payload compliance.

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9. Conclusion

The analysis concludes that the silence observed in the RTP stream was due to a mismatch between the expected and actual RTP payload sizes. Corrective actions, including proper payload size configuration and enhanced monitoring, are recommended to prevent future occurrences of this issue.

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10. 📋 Files

Wireshark Capture (.cap)