RTCP Analysis

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**1. What are the SSRC identifiers of the four RTP streams? What is your own SSRC identifier?**

Our SSRC is the one in the SENDING packet, for example the first packet: 3363156872

The SSRC of the four RTP streams are (All are from RECEIVED packets):

295596880

2574718951

2675341336

4106489149

**2. Using the information in the log file, calculate the amount with which the RTP timestamp increases each second. These are of course estimates; what do you think the true values are?**

To calculate this, I used a spreadsheet (which I’ll include).

For every RTCP received stream, I took 4 SR packets which I’ll call p1, p2, p3 and p4.

Then I calculated the difference in RTP Timestamp and NTP Timestamp for every pair (p2 - p1, p3-p2, p4-p3).

For each pair I now have a RTP delta and a NTP delta. I then divided the RTP delta by its corresponding NTP delta. (#samples between packets / time between between packets)

We now have the sampling frequency for each pair.

I proceeded by adding these frequencies and dividing them by the amount of frequencies to get an average frequency of this stream.

The results are in the table below:

|  |  |
| --- | --- |
| SSRC | Average sampling frequency (Hz) |
| 295596880 | 89999.89275 |
| 2574718951 | 7999.99901 |
| 2675341336 | 48000.07004 |
| 4106489149 | 89999.75992 |

By looking at the RTP payload types and their frequencies\*, we can deduct what the actual numbers should be:

|  |  |
| --- | --- |
| SSRC | Average sampling frequency (Hz) |
| 295596880 | 90000 |
| 2574718951 | 8000 |
| 2675341336 | 48000 |
| 4106489149 | 90000 |

\*https://en.wikipedia.org/wiki/RTP\_audio\_video\_profile

**3. Two of the four streams are video and two are audio. Which SSRCs are which?**

We know from 2) that there are 2 streams which have a sampling rate of 90KHz.

Based on the RTP payload types, only video should have a sampling rate of 90KHz. Thus streams 295596880 and 4106489149 should be video streams.

Since we already found the two video streams the other two should be audio streams, this is confirmed by their sampling rate (8000 and 48000).

**4. Which packet(s) can you use to calculate the jitter with which X receives the RTP streams?**

We need to look at the packets that we send out. In each SR packet we send out, we list the Jitter for each one of the streams. For example, in the first packet we send out, stream 2574718951 has a jitter of 4328. This value is in the time units of the RTP timestamp.

**5. Calculate the jitter for each of the four RTP streams.**

Stream 2675341336 has a jitter of 0, this means that there is absolutely no jitter in this stream, thus the jitter is 0 seconds.

Stream 2574718951 has a jitter of 4328, this is in RTP timeunits so we need to convert it to a more understandable format such as seconds. We know the sampling frequency is 8KHz so that means 8000 samples is 1 second. Thus we can calculate how many seconds 4328 units is. 4328 / 8000 = 0.541 seconds.

Stream 295596880 has a jitter of 4251 and a sampling frequency of 90KHz, 4251 / 90000 = 0.04723 seconds.

Stream 4106489149 has a jitter of 1 and a sampling frequency of 90KHz, 1 / 90000 = 0.00001 seconds.

What we have calculated is the variation in arrival time. For example, the packets of stream 2574718951 arrives 0.541 seconds too soon or too late.

**6. Estimate the packet loss with which each of the four RTP streams are received by X, based on the ‘Fraction lost’ field. What is the relevant time interval for this value?**

Based on the RFC, the fraction lost is equal to:

(#lost packets / #expected packages) \* 256

So this is the percentage of lost packages multiplied by 256 between two SR packets.

If we look at the first SR packet we send we see:

SSRC 2675341336

Fraction lost: 52

Fraction lost / 256: 0,20312 = 20%

SSRC 2574718951

Fraction lost: 0

Fraction lost / 256: 0 = 0%

SSRC 295596880

Fraction lost: 2

Fraction lost / 256: 0,00781 = 1%

SSRC 4106489149

Fraction lost: 0

Fraction lost / 256: 0 = 0%

We can see that the first stream loses 20% of its packets, the second 0%, the third 1% and the last 0%.

This is over the time-interval between two sent SR packets. In our case that would be 4,82085 seconds.

**7. For SSRC 2675341336, you receive an RTP packet with timestamp 153765211. Calculate at which time this packet was sampled according to the sender of the packet**

From the previous questions we know that this stream has a frequency of 48KHz.

We can just take a random SR packet from this stream:

NTP timestamp: 1405607977.219841  
RTP timestamp: 153353227

Delta RTP timestamp = **153765211 -** 153353227 = 411984

Because we know the frequency, we know that 1 RTP time unit equals 1 / 48000 seconds.

Thus we can calculate the wall time:

Walltime = 1405607977.219841 + (411984 \* 1/48000) =

1405607977.219841 + 8.583 =

1405607985,80284