



A note from the student behind this lab

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RTP, SRTP, and codecs

RTP version 2 carries a sequence number, a timestamp, an SSRC, and a marker bit. SIP does not carry those fields.

Version 2 and the fields beside it

RFC 3550 defines RTP, and the version field in the header is 2. The same header carries a marker bit, a 7-bit payload type, a 16-bit sequence number, a 32-bit timestamp, and a 32-bit SSRC.

The sequence number increases by one for each RTP packet that source sends, so a gap is a loss signal. The SSRC is chosen by the sender and identifies that source inside the session, which is a different job from the SIP Call-ID.

The timestamp is a media clock

The timestamp advances by the number of clock ticks in the packet, using the payload clock, and it is not a wall-clock time. For 20 ms of G.711 at 8000 Hz the timestamp increases by 160, and the same step of 160 applies to G.722 because its rtpmap clock is also 8000.

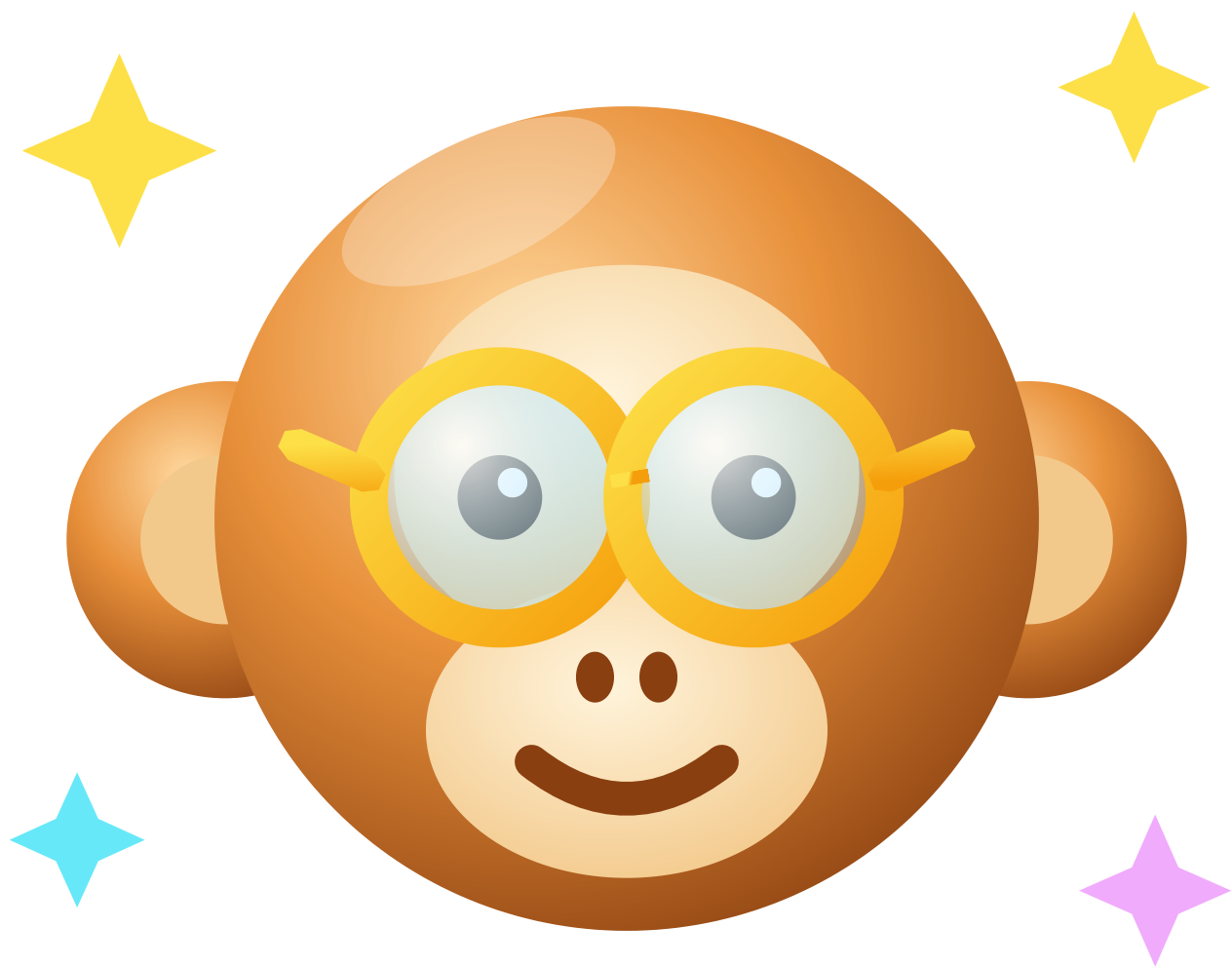
For audio, the marker bit is set on the first packet after a silence gap, which is an RFC 3551 use of the bit RFC 3550 defined. A 200 OK never contains these fields, so a media fault is read from the RTP capture.

Room: [The RTP header](#)

Sources and go deeper

- [RFC 3550](#) — last checked 2026-10
- [RFC 3551](#) — last checked 2026-10

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