

The root cause: two clocks, never the same

Two clocks	Capture and playback run on separate crystals; rates differ by parts per million (ppm).
Consumer spec	Typical crystals +/-20 to +/-50 ppm; two devices can differ by up to 100 ppm.
The squeeze	Faster producer overflows the buffer; slower producer empties it into silence.

How drift accumulates

50 ppm	= 50 / 1,000,000 = 0.00005 fractional clock difference.
per second	x 1 s = 0.05 ms gained or lost; at 48 kHz that is 2.4 samples/s.
per hour	x 3,600 s = 180 ms - at the ITU-R BT.1359-1 limit (+90 / -185 ms).

Soft vs hard correction

Soft / resample	Nudge a fraction of every sample. Inaudible. Higher CPU. The audio default.
Hard / drop+insert	Remove or repeat a whole 10-20 ms frame. Cheap. Audible click, gap, stutter.
WebRTC NetEQ	Accelerate shortens, PreemptiveExpand lengthens - pitch-preserving time stretch.
Video	Frames are discrete: drop or repeat only. The eye forgives it; the ear does not.

The audio-master rule for video

Audio master	Audio is the reference timeline; video drops or repeats frames to follow it.
Why	A 10 ms audio gap is more noticeable than a duplicated video frame.

Four common mistakes

- Blaming the network: jitter is random; drift is a steady one-directional slide.
- Reactive-only buffer: correct continuously, do not wait for empty or full.
- Pitch-shifting resampler: use pitch-preserving time stretch, not a raw rate change.
- Inverted master: drop video frames to follow audio, never the other way round.
- Tell-tale: fine at the start, worse over time = clock drift, never a one-off bug.