

The three modes, the bitrate sweet spots, the FEC/DTX/frame-size dials, and the rule that picks Opus over AAC.

## Three modes

- SILK-only** Speech engine (linear prediction). Voice at low bitrates.
- CELT-only** Music engine (MDCT). Music and lowest-latency settings.
- Hybrid** SILK on lows + CELT on highs. Natural full-band speech.

## Bitrate sweet spots (20 ms frame, RFC 7587)

- Narrowband speech **8-12 kbit/s**
- Wideband speech **16-20 kbit/s**
- Full-band speech **28-40 kbit/s**
- Full-band mono music **48-64 kbit/s**
- Full-band stereo music **64-128 kbit/s**

## The dials that matter

- Bitrate** 6 to 510 kbit/s, changeable at any instant.
- Frame size** 2.5 / 5 / 10 / 20 / 40 / 60 ms. 20 ms is the real-time default.
- FEC (useinbandfec=1)** Puts a backup of the prior packet in the next one. Recovers loss.
- DTX (usedtx=1)** Stops sending during silence; emits comfort noise.
- SDP** rate=48000, channels=2; negotiate stereo, cbr, maxaveragebitrate.

### The one rule

Live two-way conversation -> Opus. One-way playback to many devices -> AAC.  
Doing both? Opus for the calls, AAC for the recorded library.  
Enable FEC and DTX on BOTH ends, or the redundancy is sent and thrown away.