# Live Audio Contribution - Cheat Sheet

*Fora Soft Learn*

From venue to studio to viewer: AES67, SMPTE ST 2110-30, clocking & contribution codecs.

## Two journeys, not one

| | | |
|---|---|---|
| **Contribution** | Venue -> studio | High quality, low delay; re-encoded later |
| **Distribution** | Studio -> audience | Compressed, scalable; seconds of delay OK |

## Inside the facility: AES67 vs SMPTE ST 2110-30 (uncompressed PCM)

| PROPERTY | AES67 | ST 2110-30 (Level A) |
|---|---|---|
| **Audio format** | Uncompressed PCM (L16 / L24) | Uncompressed PCM (16 / 24-bit) |
| **Sample rate** | 44.1 / 48 / 88.2 / 96 kHz | 48 kHz (96 kHz higher levels) |
| **Transport** | RTP over UDP | RTP over UDP (per AES67) |
| **Clock** | PTP (IEEE 1588-2008) | PTP (IEEE 1588-2008) |
| **Packet time** | 125 us - 4 ms (1 ms default) | 1 ms; shorter in 'X' levels |
| **Channels / stream** | Flexible | 1-8 (Level A); more in B / C |

## Leaving the building: contribution over the public internet

| | | |
|---|---|---|
| **AAC-LD** | ~20 ms delay | Fraunhofer low-delay; broadcast standard |
| **AAC-ELD** | ~15 ms (7.5 ms reduced) | AAC-ELD family; two-way comms |
| **Opus** | 2.5-60 ms frames | Open; in-band FEC rebuilds lost frames |
| **Transport** | SRT / RIST / Zixi | UDP + ARQ; recovers loss, not TCP |

## Why uncompressed stays inside: bandwidth math

bandwidth = sample_rate (Hz) x bit_depth (bytes) x channels

Stereo 48 kHz / 24-bit: 48000 x 3 x 2 = 2.304 Mbps.

8-channel 48 kHz / 24-bit: 48000 x 3 x 8 = 9.216 Mbps.

Fine on a 1/10 GbE LAN; impossible to guarantee over the open internet.

## Remember

- [ ] Two trips: contribution (quality + low delay) then distribution (compressed + scalable). Plan them separately.
- [ ] AES67 = uncompressed PCM over RTP so multi-vendor gear interoperates; ST 2110-30 references AES67 + adds levels.
- [ ] ST 2110-30 Level A is mandatory: 48 kHz, 16/24-bit, 1-8 channels, 1 ms packets. Use -31 to tunnel AES3 / Dolby E.
- [ ] PTP gives one shared clock: 48,000 samples = 1 second on every device. Drift = a clocking bug, not a codec bug.
- [ ] Leaving the building? Low-delay codec (AAC-ELD / Opus) + reliable transport (SRT first choice in 2026).
- [ ] EBU Tech 3326 (ACIP): G.711/G.722/MPEG L2/PCM required; AAC + AAC-LD recommended; Opus optional (2014 list).