

Live Audio Contribution - Cheat Sheet

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

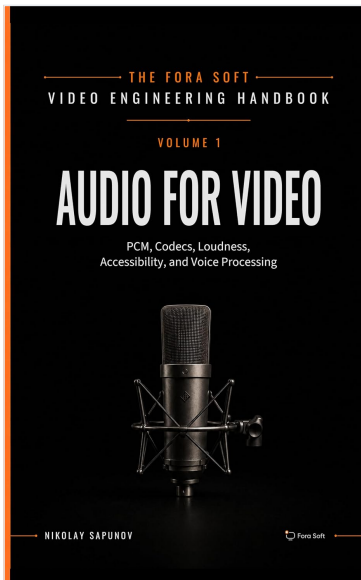
AAC-LD	~20 ms delay	Fraunhofer low-delay; broadcast standard
AAC-ELD	~15 ms (7.5 ms reduced)	AAC-ELD family; two-way comms
Opus	2.5-60 ms frames	Open; in-band FEC rebuilds lost frames
Transport	SRT / RIST / Zixi	UDP + ARQ; recovers loss, not TCP

Why uncompressed stays inside: bandwidth math

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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.
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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).



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Audio for Video: A Complete Guide to Sound in Video Products

From PCM and Codecs to Loudness, Accessibility, and Real-Time Voice Processing

By Nikolay Sapunov, CEO at Fora Soft · 1,229 pages · Kindle & paperback on Amazon

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This cheat sheet compares AES67 and ST 2110-30 inside the facility and the codecs and transports for leaving it. Volume 4 covers the audio material in depth: sampling, bit depth and channel layouts, loudness, codecs such as Opus and AAC, and real-time voice processing.

WHAT YOU WILL FIND INSIDE

- Sampling, bit depth and bandwidth math
- Channel layouts and multichannel audio
- Codecs: Opus, AAC and speech codecs
- Loudness and LUFS targets for broadcast

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Planning remote contribution or an AES67 / ST 2110 audio workflow?

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