2wcom Audio Codec Matrix – Broadcast & Streaming Workflows

Codec	Bitrate range (kBit/s)	Optimum/typical stereo bitrate (kBit/s)	Algorithmic delay (ms)	Compression Technology	Strength	Application areas
MPEG-1 Layer 2	32–384	192–384	40–60	Filterbank + simple psychoacoustic model	Low complexity, fast decode	Broadcast, DVB, DAB
MPEG-1 Layer 3 (mp3)	32–320	128–320	100–120	Hybrid filterbank + MDCT, Huffman coding, joint stereo, psychoacoustic masking	High compatibility, decent quality	Web streaming, audio player, broadcast contribution/distribution
AAC-LC	16–320	96–256	40–60	Pure MDCT, TNS (Temporal Noise Shaping), PNS (Perceptual Noise Substitution), Huffman	Better coding efficiency, better quality at lower bitrates	Streaming, iTunes, video containers, broadcast contribution/distribution
HE-AAC	16–128	48–96	80–120	AAC LC + SBR (Spectral Band Replication)	Efficient coding at low bitrates, preserving high frequencies	Mobile streaming, DVB, DAB+
HE-AAC-v2	16–64	32–48	80–120	HE-AAC + Parametric Stereo (PS)	Stereo audio at very low bitrates	Mobile streaming, DVB, DAB+

xHE-AAC	6–500	12–96	20–60	HE-AAC v2 + speech coding, improved SBR/PS + dynamic switching + loudness metadata	Handles music/speech seamlessly, acceptable quality at extremely low bitrates	Adaptive streaming, Digital Radio Mondiale (DRM), VoIP
AAC-LD	32–128	64–96	20–40	MDCT, optimzed for short framelengths	Low latency with good quality	Video conferencing, broadcast contribution
AAC-ELD	24–128	48–64	20–32	AAC-LD + Spectral Band Replication (SBR)	Better quality at same latency	Broadcast contribution, Intercom
AAC-ELD-v2	24–128	48–64	24–36	AAC-ELD + Parametric Stereo (PS)	Stero at very low bitrates & delay	Broadcast contribution, Intercom
Opus	6–510	32–128	5–30	SILK + CELT (MDCT), frame-by-frame switching, variable frame size	Low latency, robust voice/music mix	VoIP, WebRTC, broadcast contribution/distribution
aptX	352	352	150–200	ADPCM (Adaptive Differential Pulse Code Modulation), fixed compression ratio	Simple, low complecxity	Optimized for 4:1 compression ratio (4x stereo over AES3/ISDN)
Enhanced aptX	384–576	384–576	< 5ms	Non-framed ADPCM	Very low delay	Remote and STL links
FLAC	> 300	500–1100 (variable)	5–10 ms	Predictive coding + Rice entropy coding (lossless)	Bit-exact quality, open format, fast decoding	Archiving, Hi-Res audio, streaming (Tidal, Qobuz)

All codecs shown are supported across 2wcom's hardware and software portfolio.