Digital Signal Processing for Music

Part 24: Redundancy Coding

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redundancy coding

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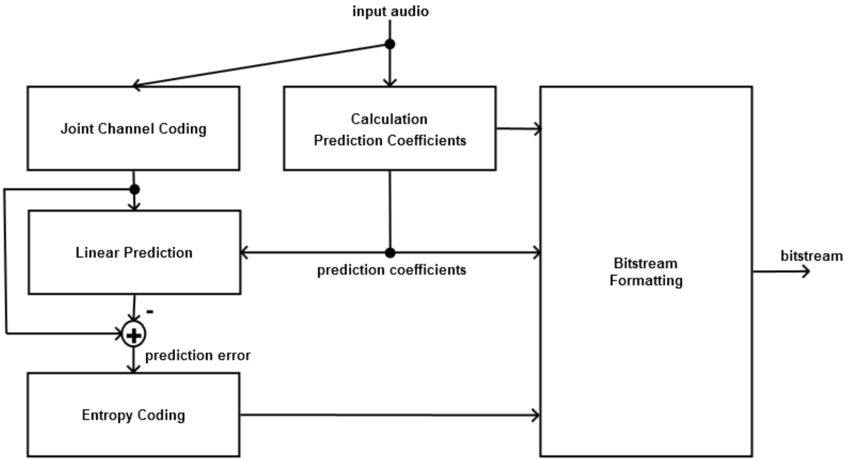

redundancy coding
overview 2/2

- **properties**
  - perfect signal reconstruction
  - bitrate reduction depends on input signal
    - typical gain (stereo, 48k): factor 2
  - no constant bitrate $\rightarrow$ streaming only with large buffers

- **common applications/algorithms**

name	sampling rates	channels	word length
Shorten	all	2	8/16
FLAC	1-1048k	8	4-32
MLP	44.1k-192k	63	1-24
ALS	all	65536	1-32 (int), 32(float)

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