Digital Signal Processing for Music

Part 24: Source Coding

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Intro

>> Typical audio bit rates

$$16] {
m bit} \cdot 44100] {
m sps} \cdot 2] {
m chan} = 1411.2] {
m kbps}$$
 $24] {
m bit} \cdot 192000] {
m sps} \cdot 5] {
m chan} = 23040] {
m kbps}$

- >>> Reasons for bit rate reduction
 - >> Economical reasons: Cheaper transmission/storage
 - >> Technical reasons: Restricted storage / transmission bandwidth
- >> Applications for source coding
 - >> Internet: streaming, distribution, p2p, VoIP, ...
 - >> Media: DVD-V/A, ...

Reducing Bitrate

>> Lossless:

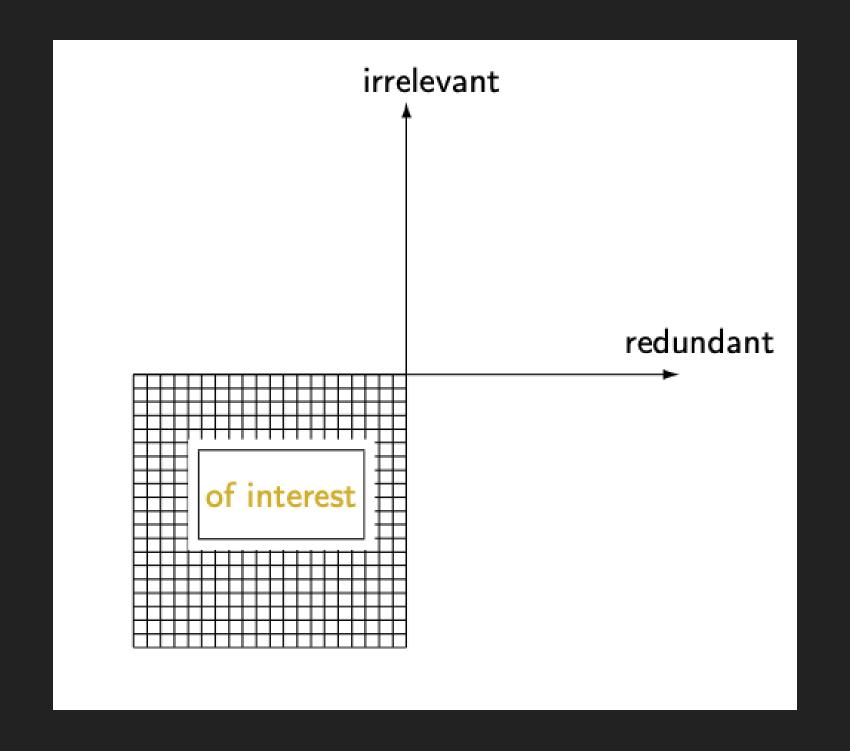
Remove *redundant* information (unnecessary to reconstruct the signal)

- >> Entropy coding
- >> (Linear predictive coding)

>> Lossy:

Remove *irrelevant* information (not "missed" by the recipient)

- >> Waveform coding
- >>> Perceptual coding



Information Theory: Definitions

Note: Words to be transmitted referred to as symbols

Information Content

The less frequent a symbol, the higher its information content, self-information, surprisal

$$I_n = \log_2\left(rac{1}{p_n}
ight)$$



Information Theory: Definitions

Entropy

The Expected Value of the information content; the theoretic minimum of bits required for transmission

$$H = \sum_{n=0}^{N-1} p_n \cdot I_n$$



Information Content & Entropy Examples

Dice:
$$p_n = \frac{1}{6}$$

$$I_n = \log_2 \left(rac{1}{p_n}
ight) = 2.58 \mathrm{bit}$$
 $H = 2.58 \mathrm{bit}$

Imperfect dice:
$$p_0=rac{1}{2},\;p_{1\dots 5}=rac{1}{10}$$

$$egin{aligned} I_1 &= \log_2{(2)} &= 1 ext{bit} \ I_{2\ldots 6} &= \log_2{(10)} &= 3.32 ext{bit} \ H &= rac{1}{2} \cdot 1 + rac{5}{10} \cdot 3.32 = 2.16 ext{bit} \end{aligned}$$



Entropy Coding

Idea: Use shorter words for frequent symbols

3 possible signals

Symbol	Probability	Word
A	p = 0.5	0
В	p = 0.25	10
С	p = 0.25	11

>> Entropy

$$H = \sum_{n=0}^{N-1} p_n \log_2\left(rac{1}{p_n}
ight) = 1.5$$

- >> Transmit the following group of symbols: ABCA > 010110
- >> Required bits:

$$rac{transmitted\ bits}{transmitted\ symbols} = rac{6}{4} = 1.5$$

Entropy Coding

>> 3 possible signals

Symbol	Probability	Word
A	p = 0.7	0
В	p = 0.2	10
С	p = 0.1	11

>> Entropy

$$H=\sum_{n=0}^{N-1}p_n\log_2\left(rac{1}{p_n}
ight)=1.11$$

- >> Transmit the following group of symbols: $ABCA \rightarrow 010110$
- >> Required bits:

$$rac{transmitted\ bits}{transmitted\ symbols} = rac{6}{4} = 1.5$$

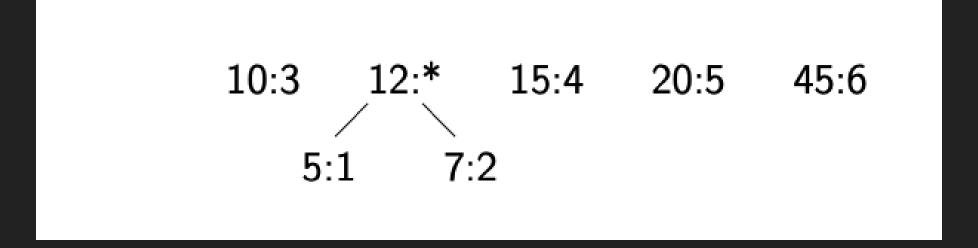
Huffman Coding

Steps

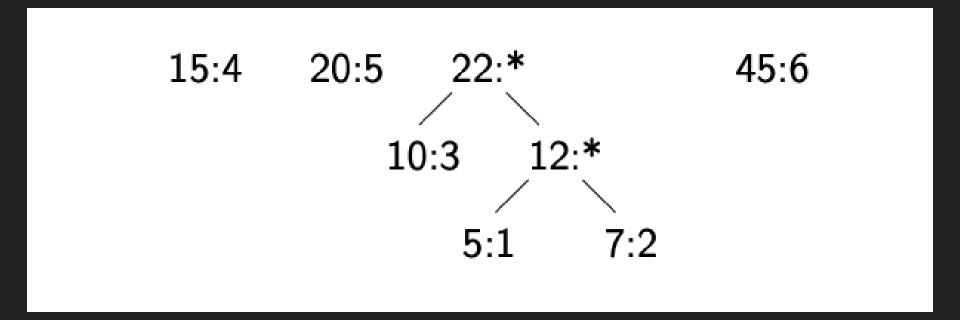
- >> Sort symbols according to frequency
- >> Combine two lowest symbols into new entry (sum)
- >> Add new entry to list
- >> Repeat until only one element left

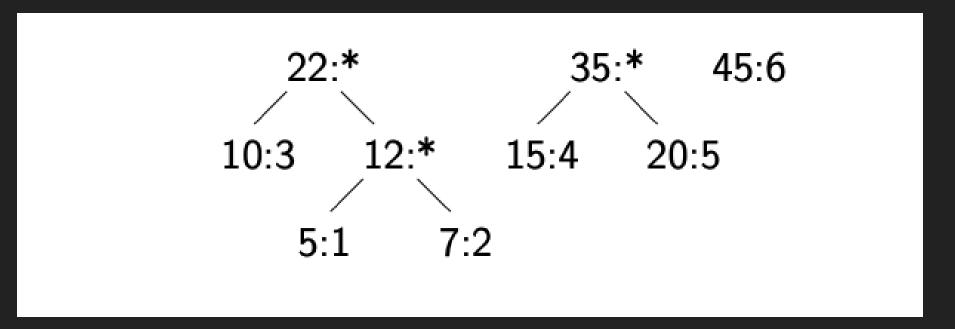
5:1 7:2 10:3 15:4 20:5 45:6

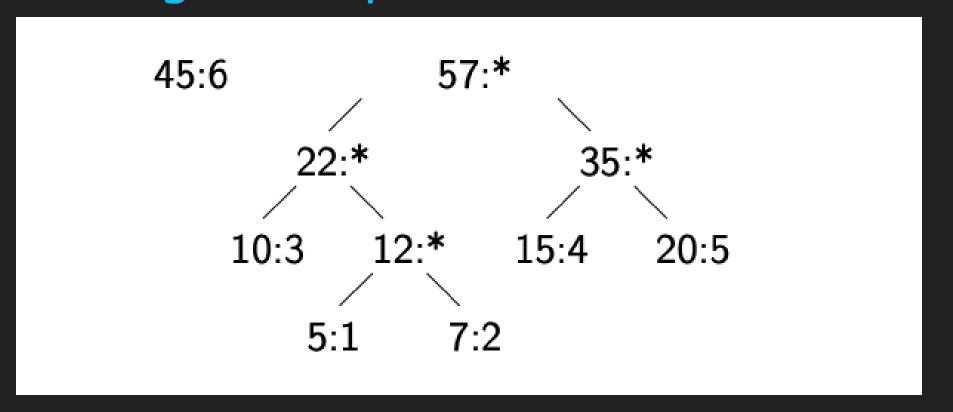


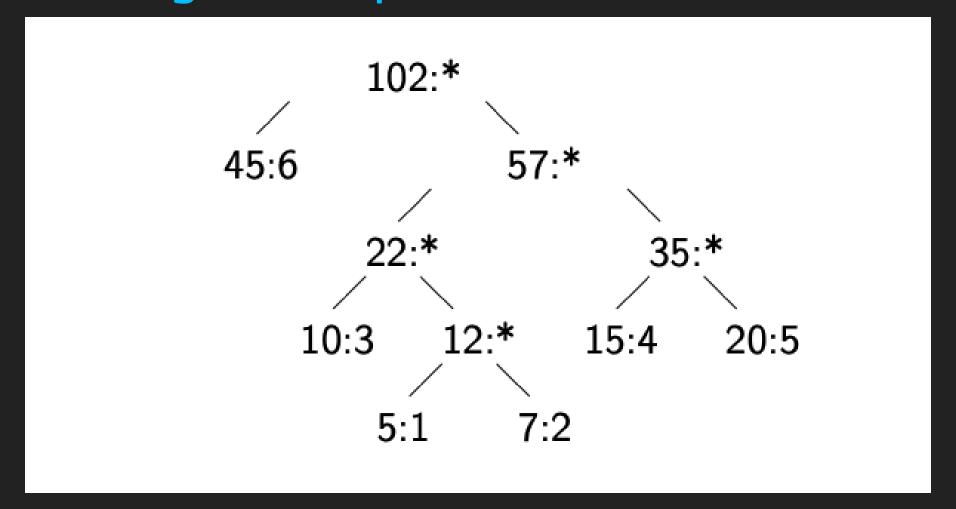




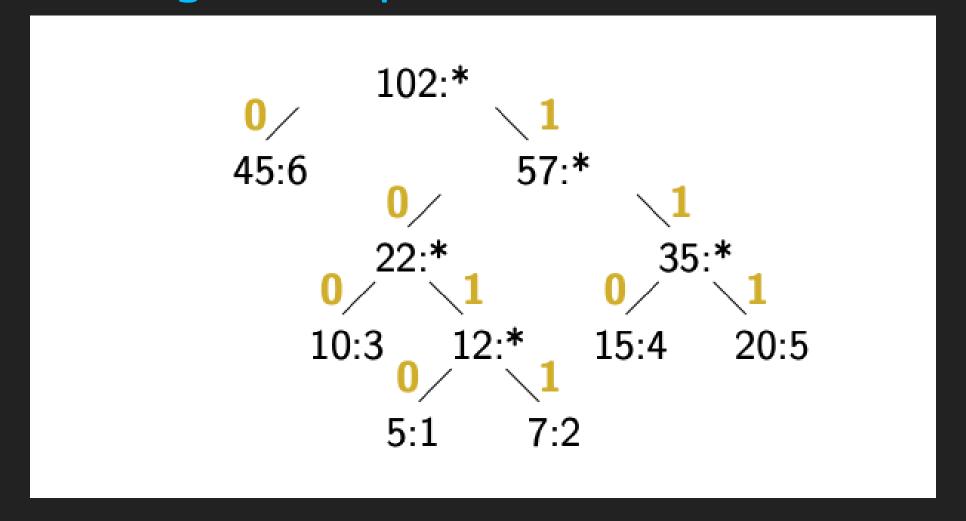




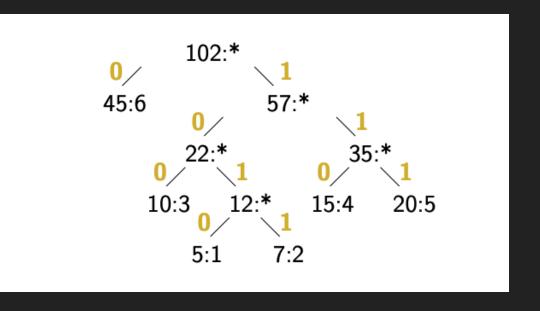








Frequency	Symbol	Code
5	1	1010
7	2	1011
10	3	100
15	4	110
20	5	1111
45	6	O



Huffman Coding for audio signals

- \Rightarrow Symbols: 2^w
- >> PDF indicates probability per symbol

Arithmetic Coding

- Huffman coding is only optimal if $p_n=1/2^k$ Alternative: **Arithmetic Coding**
- >> Allows other probability distributions
- \blacktriangleright Encodes the whole sequence in one fractional number $0.0 \leq f < 1.0$

Steps

Arithmetic Coding: Decoding Example

Sequence ABCA, $p_A = 0.6$, $p_B = 0.2$, $p_C = 0.1$, $p_T = 0.1$ A = [0, 0.6), B = [0.6, 0.8), C = [0.8, 0.9), T = [0.9, 1) **Decoding** 0.463



Arithmetic Coding: Encoding Example

Sequence ABCA, $p_A = 0.6$, $p_B = 0.2$, $p_C = 0.1$, $p_T = 0.1$ A = [0, 0.6), B = [0.6, 0.8), C = [0.8, 0.9), T = [0.9, 1) **Encoding** 0.463

- >> Select segment 1, set interval to [0, 0.6)
- >> Select segment 2, set interval to [0.36, 0.48)
- >> Select segment 3, set interval to [0.456, 0.468)
- >> Select segment 1, set interval to [0.456, 0.4632)
- >> Select segment 4, set interval to [0.46248, 0.4632)
- >> Choose value from last segment (e.g., 0.463) and transmit

Fundamentals: Linear Prediction

Idea: Use preceding samples to estimate/predict future samples

Fundamentals: Linear Prediction - First Order Prediction

- \Rightarrow Prediction: $\hat{x}(i) = b_1 \cdot x(i-1)$
- >> Prediction error:

$$egin{aligned} \sigma_e^2 &= \mathcal{E}ig\{(x(i)-b_1x(i-1))^2ig\} \ &= \sigma_x^2 + b_1^2\sigma_x^2 - 2b_1
ho_{xx}(1) \ &= ig(1+b_1^2-2b_1
ho_{xx}(1)ig)\sigma_x^2 \end{aligned}$$

Fundamentals: Linear Prediction - First Order Prediction

>> Optimum coefficient: $\frac{\partial \sigma_e^2}{\partial b_1} = 0$

$$egin{aligned} 2b_1\sigma_x^2 - 2
ho_{xx}(1)\sigma_x^2 &= 0 \ b_1 &=
ho_{xx}(1) \end{aligned}$$

>> Minimum prediction error power:

$$egin{align} \sigma_e^2 &= ig(1+b_1^2-2b_1
ho_{xx}(1)ig)\sigma_x^2 \ &= ig(1+
ho_{xx}(1)^2-2
ho_{xx}(1)
ho_{xx}(1)ig)\sigma_x^2 \ &= ig(1-
ho_{xx}(1)ig)\sigma_x^2 \ \end{gathered}$$

$$\sigma_e^2 = (1-
ho_{xx}(1))\sigma_x^2$$



Linear Prediction - Coefficients

- >> Prediction gain depends on
 - \blacktriangleright Predictor coefficients b_j
 - >> Signal
- >> Optimal coefficients can be derived by finding minimum of prediction error

$$egin{aligned} rac{\partial \sigma_e^2}{\partial b_j} &= 0 \ r_{xx}(\eta) &= \sum_{j=1}^{\mathcal{O}} b_{j, ext{opt}} \cdot r_{xx}(\eta-j), & 1 \leq \eta \leq \mathcal{O} \ ec{r}_{xx} &= R_{xx} \cdot ec{b}_{ ext{opt}} \ ec{b}_{ ext{opt}} &= R_{xx}^{-1} \cdot ec{r}_{xx} \end{aligned}$$

Linear Prediction - Summary

>> Predictor Length

- >>> Rule of thumb: the longer the predictor, the better the prediction
- >> Can range from 10 coefficients to hunders

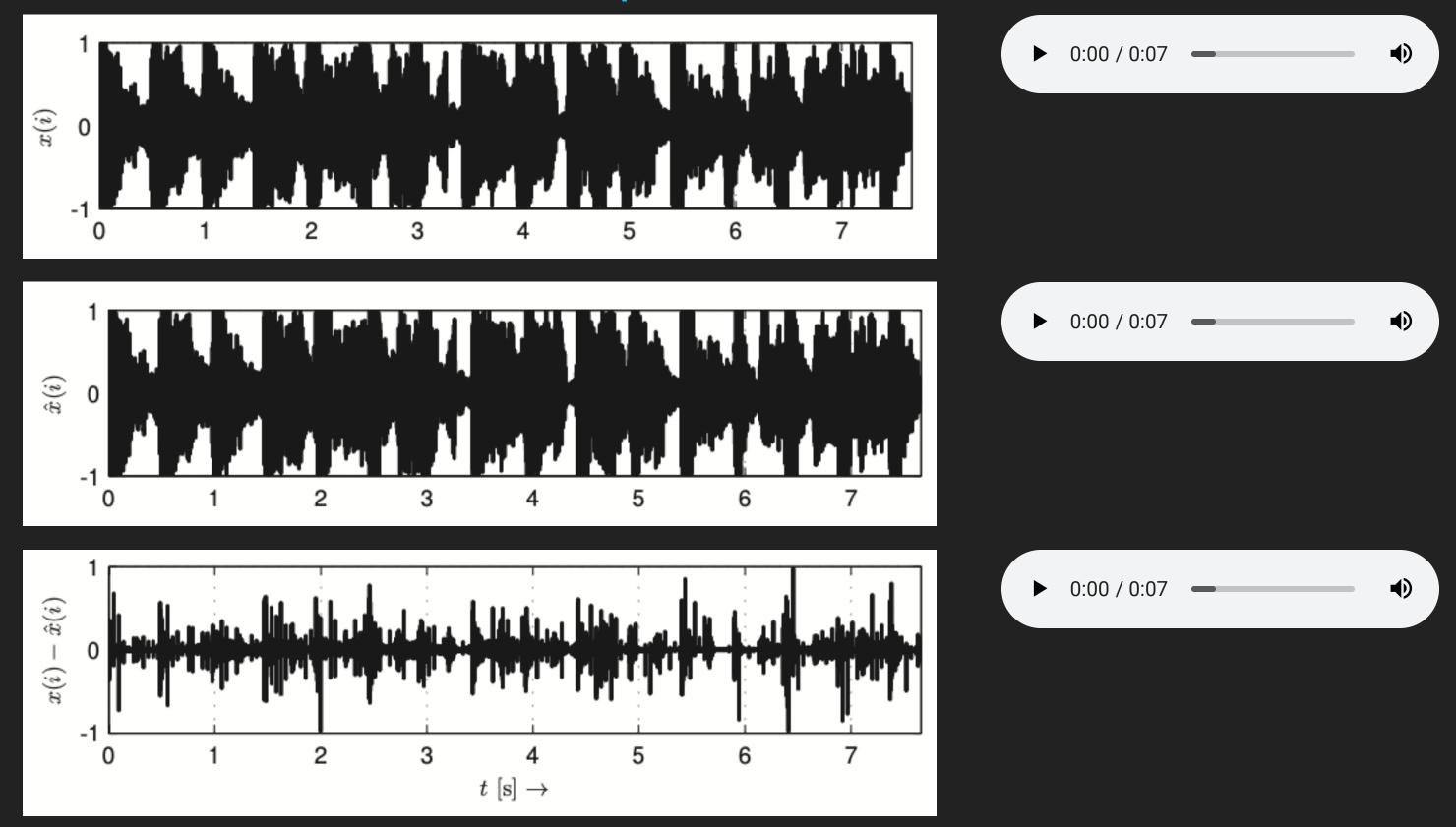
>> Predictor coefficient updates

>>> Better signal adaptation if coefficients are updated block-by block

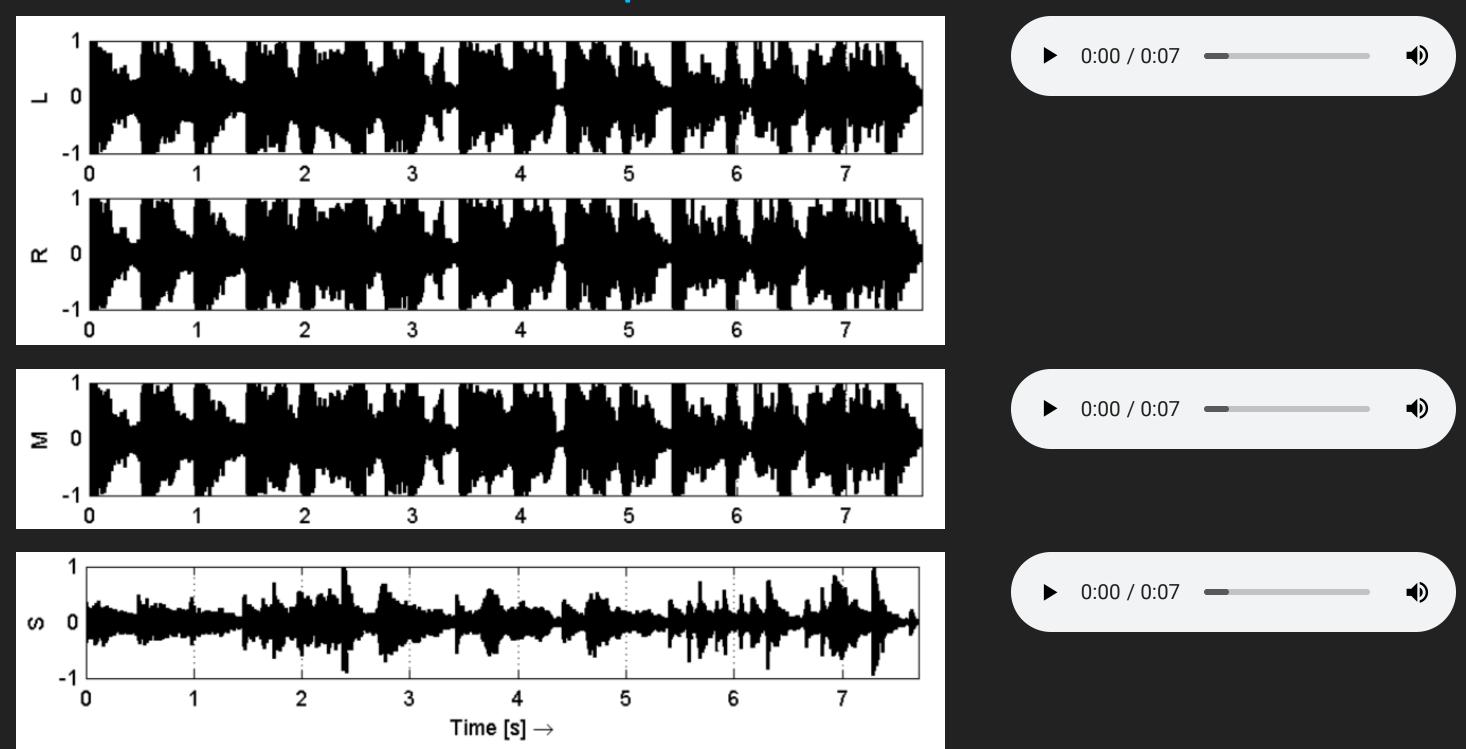
>> Input Signals

- >> White noise / random processes cannot be predicted
- >> Periodic signals may theoretically be perfectly predicted

Linear Prediction Examples



Linear Prediction Examples

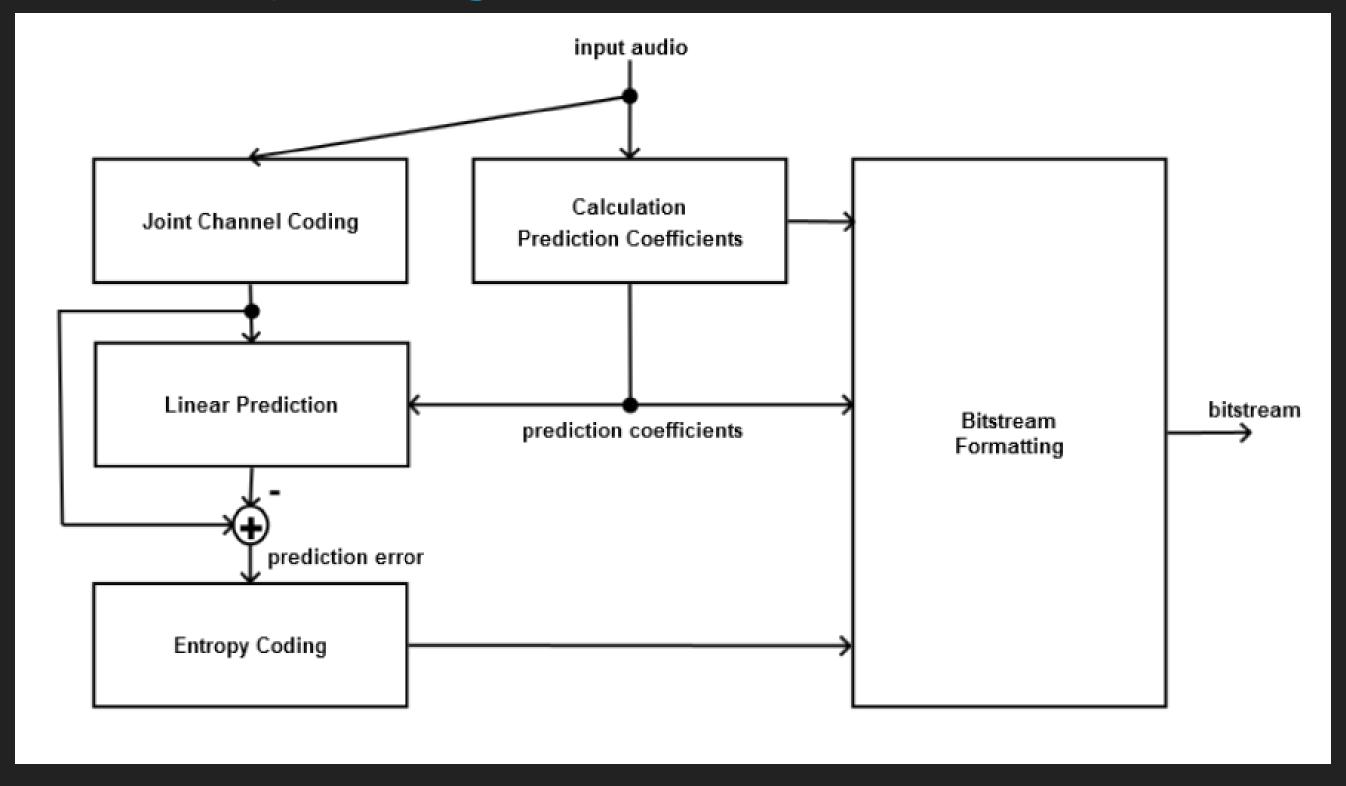




Summary

- >>> Bitrate can be reduced by removing redundancy and/or irrelevance
- >> Removing redundancy:
 - >>> Entropy Coding: Transmit frequent symbols with shorter codes
 - >>> Linear Prediction: Transmit diff signal plus predictor coefficients
- >>> Removing irrelevance:
 - >> Reduce quantization wordlength / lower sample rate
 - >> More techniques discussed in future classes

Redundancy Coding



>> Properties

- >>> Perfect signal reconstruction
- >>> Bitrate reduction depends on input signal Typical gain (stereo, 48k): Factor 2
- >> No constant bitrate -> Streaming only with large buffers

>>	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)