

Optimization of time-frequency transforms for audio coding using a perceptive measure of distortion and a sparsity constraint

Ichrak Toumi & Olivier Derrien - LMA

FLAME Meeting, November 2014







1. Compression of audio signal and time-frequency transforms

◆□▶ ◆□▶ ◆ □▶ ◆ □▶ ─ □ ─ のへぐ

- > 2. Writing compression as a standard optimization problem
- ► 3. Choosing suitable time-frequency transforms
- 4. Preliminary results with union of MDCTs
- ► 5. Perspectives

Definition of audio coding

- Minimize the amount of information to be stored/transmitted for near-perfect audio quality
- or maximize the audio quality for a given amount of information

Link with time-frequency transforms

- State-of-the art codec (MP3, AAC) rely on invertible time-frequency transforms (PQMF-Banks, MDCT)
- because audition can be efficiently modelized in the TF domain

Sparsity in audio coding

- Reducing the amount of information is achieved by
 - Setting some coefficients to zero
 - Re-quantize non-zero coefficients
- Example: AAC @ 128 kbps
 - ▶ Stored/transmitted information: 10% of the original
 - ▶ Non-zero coefficients: 30% of the original
 - Remaining 20%: re-quantization
- A sparse representation is desirable for audio coding
- ► Target sparsity value : 30% non-zero coefficients or less

2. Writing compression as a standard optimization problem

► Consider discrete-time, N samples long, real-valued signals: $\mathbf{x} \in \mathbb{R}^N$

The coding transform

- Consider a time-frequency transform characterized by
- An analysis dictionary: $\mathbf{A} = \{\phi_1^H \cdots \phi_M^H\}, \phi_m \in \mathbb{C}^N$
- A synthesis dictionary: $\mathbf{S}^T = \{\psi_1^T \cdots \psi_M^T\}, \psi_m \in \mathbb{C}^N$
- ► The analysis operator is: $\mathbf{y} = \mathbf{x} \mathbf{A} \quad \Leftrightarrow \quad y_m = \langle \mathbf{x}, \phi_m \rangle \quad \forall m$

- The synthesis operator is: $\hat{\mathbf{x}} = \mathbf{y} \mathbf{S} \quad \Leftrightarrow \quad \hat{\mathbf{x}} = \sum_m y_m \psi_m$
- Perfect reconstruction \Leftrightarrow **A S** = **I**_N, which implies $M \ge N$

The perceptual transform

- ► A relevant measure for perceived distortion can be computed using a perceptual time-frequency transform of size Q ≥ M
- The analysis dictionary is: $\mathbf{P} = \left\{ \mathbf{p}_1^H \cdots \mathbf{p}_Q^H \right\}, \mathbf{p}_q \in \mathbb{C}^Q$
- There is no need for a synthesis dictionary
- ► We assume that perceptual weights µ_q > 0 associated to each vector **p**_q can be computed using an audition model

The perceptual distortion measure

$$D_p = \parallel (\mathbf{x} - \hat{\mathbf{x}}) \mathbf{P} \Delta_{\mu} \parallel^2$$

with $\Delta_{\mu} = \text{diag}(\mu_1, \cdots, \mu_Q) \Rightarrow D_p = \text{weighted L2 norm of the error}$

2. Writing compression as a standard optimization problem

Re-writing the perceptual distortion measure

$$D_p = \parallel (\mathbf{x} \, \mathbf{P} - \mathbf{y} \, \mathbf{SP}) \, \Delta_{\mu} \parallel^2$$

Formulating the coding problem

- Find **y** that minimizes D_p
- ► If we consider the quantization of y_m, y is searched only in a finite subset of ℝ^K. That will not be considered for the moment
- ► This is a weighted-L2 optimization problem of the form:

$$\operatorname{Argmin}_{\mathbf{y}}\left[\parallel\left(\mathbf{g}-\mathbf{y}\,\mathbf{K}\right)\Delta_{\mu}\parallel^{2}\right]$$

• where $\mathbf{K} = \mathbf{SP}$ is called the mixture matrix (size $M \times Q$)

Finding solutions to the coding problem

- ► The existence of solutions mainly depends on the properties of **K**
- If $rk(\mathbf{K}) = M$, the solution is unique: $\tilde{\mathbf{y}} = \mathbf{g} \mathbf{K}^{\dagger}$
- Otherwise, there is an infinite set of equivalent solutions
- ► For selecting "the best" solution, or when **K** is badly conditioned, one usually add a regularization term that promotes sparsity:

$$\operatorname{Argmin}_{\mathbf{y}}\left[\parallel\left(\mathbf{g}-\mathbf{y}\:\mathbf{K}\right)\Delta_{\mu}\parallel^{2}+\lambda\parallel\mathbf{y}\parallel^{p}\right]$$

Finding a sparse solution is especially desirable in audio coding

3. Choosing suitable time-frequency transforms

Choosing a perceptual transform: P

- Constrained by the existence of an earing model to compute μ_q
- DFT or MDCT: work with standard MPEG hearing models
- Constant-Q or ERBLett: more sophisticated models available

Choosing a coding transform: A and S

- Audio signals should naturally have sparse representations in the transform domain
- Perfect reconstruction is not necessary
- The choice shall depend on the rank of $\mathbf{K} = \mathbf{SP}$
- ► If rk(K) ≪ M, there are many local minima and the practical solution strongly depends on the initialization

- コン・4回ン・4回ン・4回ン・4回ン・4日ン

• A good choice corresponds to $rk(\mathbf{K}) \simeq M$

3. Choosing suitable time-frequency transforms

Solutions that work

- $\mathbf{A} = \mathbf{P}$ is a single MDCT Then $\mathbf{S} = \mathbf{A}^T$ and $\mathbf{K} = \mathbf{I}_M \Rightarrow$ the problem is diagonal This is a trivial case: the solution is obtained by thresholding \mathbf{g}
- ► $\mathbf{A} = \mathbf{P}$ is union of MDCTs with different sizes Then $\mathbf{S} = \mathbf{A}^T$ and $\mathbf{AS} \neq \mathbf{I}_N \Leftrightarrow$ no perfect reconstruction $\mathbf{K} \neq \mathbf{I}_M$ and $\mathrm{rk}(\mathbf{K}) < M \Leftrightarrow$ many local minima But \mathbf{K} is a very sparse matrix: when thresholding very small values to zero we get $\mathrm{rk}(\mathbf{K}) = M$

Solutions that does not work (for the moment)

- A is a MDCT and P is an ERBLett $rk(\mathbf{K}) \ll M$ and the problem can not be regularized properly
- But things seem to get better with the real part of an ERBLett

・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・
・

Analysis/synthesis matrix

- ▶ We choose the union of 2 MDCTs: 1024 bands and 128 bands
- ▶ idem AAC, but here both MDCTs can be used simultaneously
- For plots, we choose $N = 4096 \Rightarrow M = 6144$



Perceptual matrix and mixture matrix

- We assume $\mathbf{P} = \mathbf{A} = \mathbf{S}^T \Rightarrow Q = M$
- Then $\mathbf{K} = \mathbf{S} \, \mathbf{S}^T \Rightarrow K(m, q) = \langle \phi_m, \phi_q \rangle$

•
$$\operatorname{rk}(\mathbf{K}) = N = 4096 < M = 6144$$



▲□▶ ▲□▶ ▲ □▶ ▲ □▶ ▲ □ ● ● ● ●

Thresholding the mixture matrix

- We set a threshold T so that $K(m,q) \mapsto 0$ if K(m,q) < T
- This implies an error on the estimation of the distortion D_p
- ► $T = -108 \text{ dB} \Rightarrow SNR = 110 \text{ dB}$ (near perfect) and rk (**K**) = M



▲□▶▲□▶▲□▶▲□▶ ■ のへで

Thresholded mixture matrix

- ► T = -108 dB
- rk (K) = M ⇒ there is a unique solution to the optimization problem, i.e. the approximation of D_p is convex



Implementations details

- The perceptual weights μ_q are computed for both resolutions (1024 and 128 bands) with the MPEG #2 hearing model
- ► The target g = x P is computed using a standard MDCT implementation
- The thresholded mixture matrix is stored as a sparse matrix
- The signal is divided in macro-blocks, and the optimization is performed independently on each macro-block

- コン・4回シュービン・4回シューレー

- ► No redundancy is added when macro-blocks overlap
- The sparsity level is set by the regularization constant λ

Sparsity rate = 43 %

SVega original signal

SVega Reconstructed signal











Sparsity rate = 66 %

SVega original signal

SVega Reconstructed signal











Sparsity rate = 83 %

SVega original signal

SVega Reconstructed signal











- ► Try different time-frequency transforms for **S** and **P** in order to find the couple which offers the best tradeoff between perceived audio quality and sparsity rate
- Try a more sophisticated perceptive model, different from the MPEG #2

▲□▶ ▲□▶ ▲ □▶ ▲ □▶ ▲ □ ● ● ● ●

Include the quantization step in the optimization algorithm