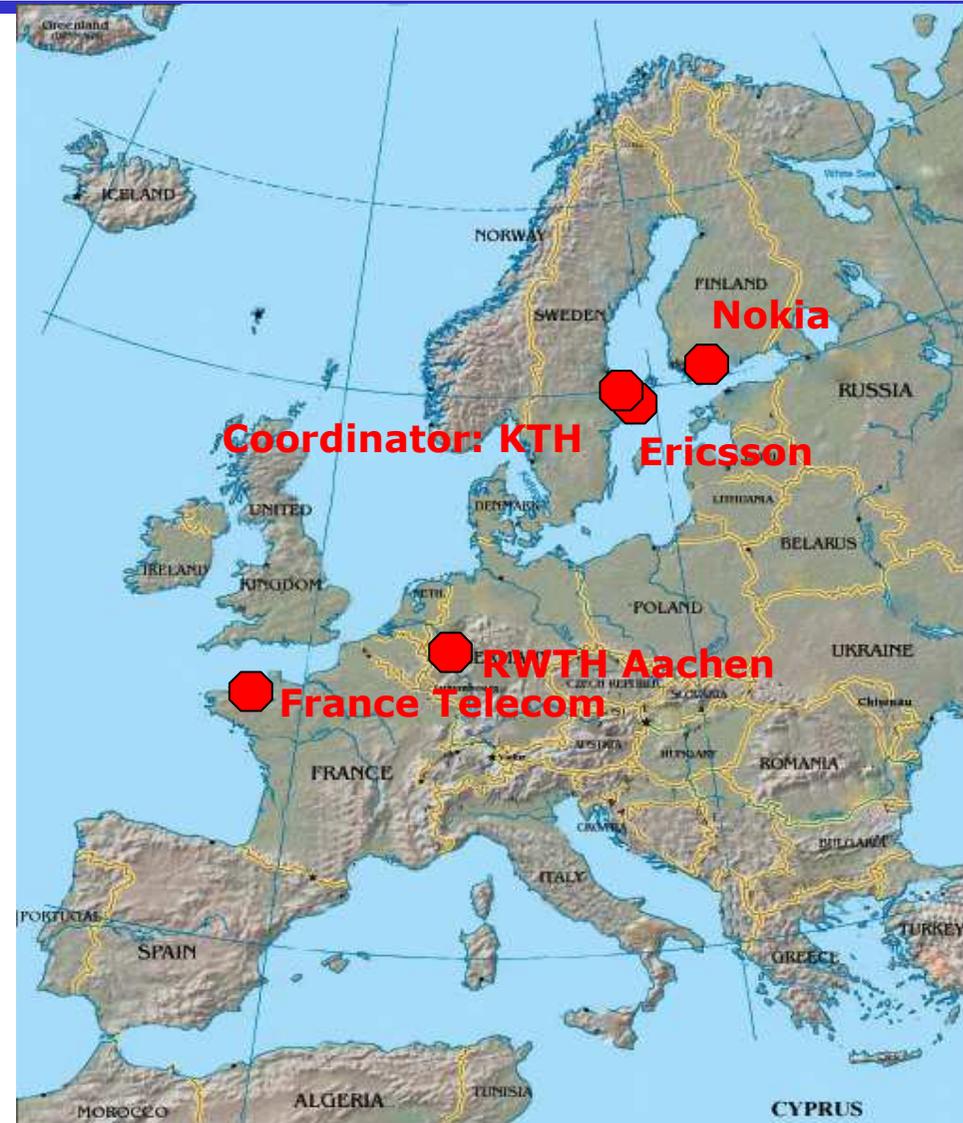
A map of Europe with several blue dots marking specific locations: one in Sweden, one in Germany, and one in France. The text 'Recent Advances in Model-based Transform Audio Coding' is overlaid in red on the map.

# Recent Advances in Model-based Transform Audio Coding

Marie Oger, Stéphane Ragot  
Flexcode Seminar Stockholm  
October 17, 2007

*FlexCode*

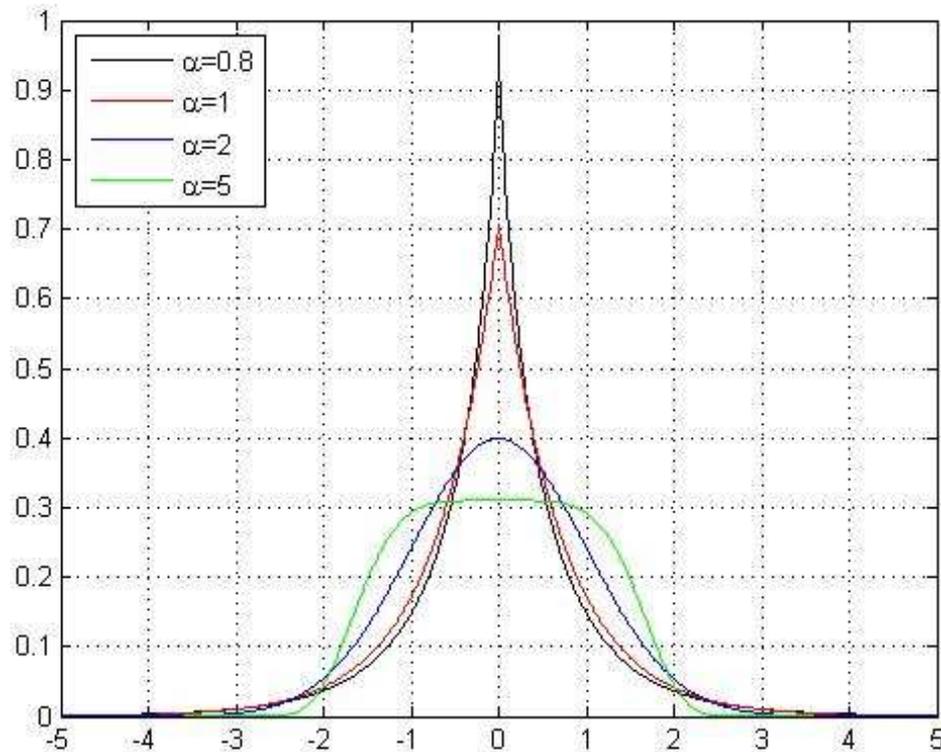
- Flexible Coding for Heterogeneous Networks
- Objectives: develop **flexible source-channel coding algorithms**
  - More flexible than current, application-specific coders
  - Flexibility through online design, **generic source, channel and distortion models**
  - Focus on audio
- <http://www.flexcode.eu>



- Approach
  - Start with existing models: transform and linear-predictive coding
  - "Flexcodize": analytic solutions → adaptive coding
- Tools for flexible coding include
  - High-rate quantization theory
  - Probability models (GMM, ...) for quantizer design
    - Quantizer specification by equations
    - Estimate statistics for source
  - Distortion measures using sensitivity matrix

- GMM (Gaussian Mixture Model) based LPC quantization [[Subramaniam 01](#)] [[Samuelsson 01](#)]
  - LPC coefficients or prediction error are modeled by a GMM
  - A mean-removed Karhunen-Loeve transform (KLT) and normalization by standard deviations is applied to LPC coefficients
- Autoregressive GMM for speech coding [[Samuelsson 04](#)]
  - Companded GMM for vector quantizers (CGMM-VQ)
- Generalized Gaussian model for image coding [[Parisot 03](#)]
  - Wavelet coding for image (EBWIC Coder)
  - Wavelet coefficients are modeled by a generalized Gaussian model

- Generalized Gaussian model
  - Definition
  - Example
- Proposed stack-run coding with model-based deadzone
  - Principle of stack-run coding
  - Rate control based on asymptotic bit allocation
  - Model-based optimization of deadzone
  - Objective & Subjective results
  - Delay & Complexity
  - Audio samples
- Latest developments: model-based bit plane coding
  - Principle
  - Preliminary results
- Conclusion & perspectives



- The probability density function (pdf) of a zero-mean **generalized Gaussian variable**  $z$  of standard deviation  $\sigma$  is given by :

$$p_{\alpha,\sigma}(z) = \frac{A(\alpha)}{\sigma} e^{-|B(\alpha)z/\sigma|^\alpha}$$

where

$$A(\alpha) = \frac{\alpha B(\alpha)}{2\Gamma(1/\alpha)} \quad \text{and} \quad B(\alpha) = \sqrt{\frac{\Gamma(3/\alpha)}{\Gamma(1/\alpha)}}$$

with  $\Gamma(\cdot)$  the Gamma function defined as

$$\Gamma(\alpha) = \int_0^\infty e^{-t} t^{\alpha-1} dt$$

- The method used to estimate  $\alpha$  is proposed by Mallat [Mallat 89]

$$F(\alpha) = \frac{E(|z|)}{\sqrt{E(z^2)}} = \frac{\Gamma(2/\alpha)}{\sqrt{\Gamma(1/\alpha)\Gamma(3/\alpha)}}$$

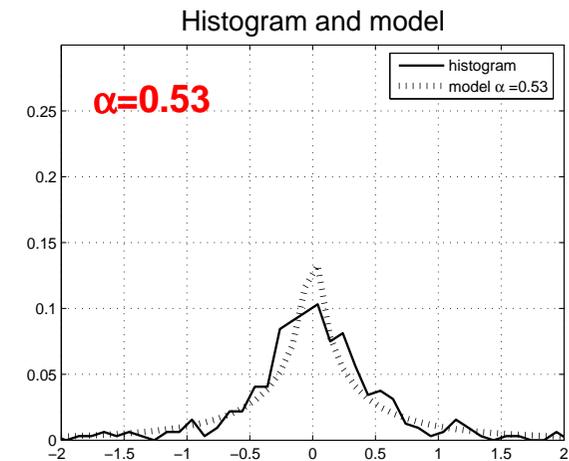
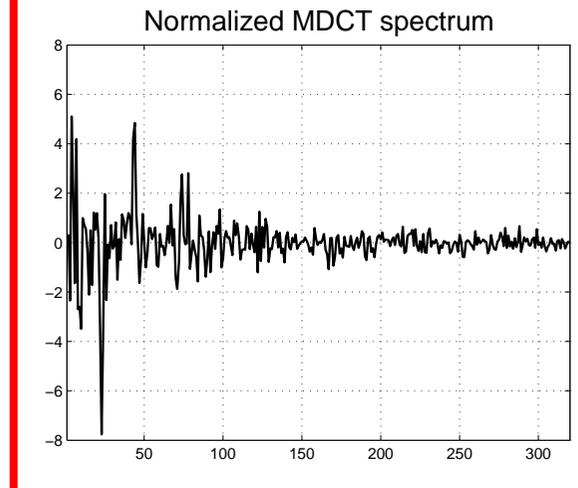
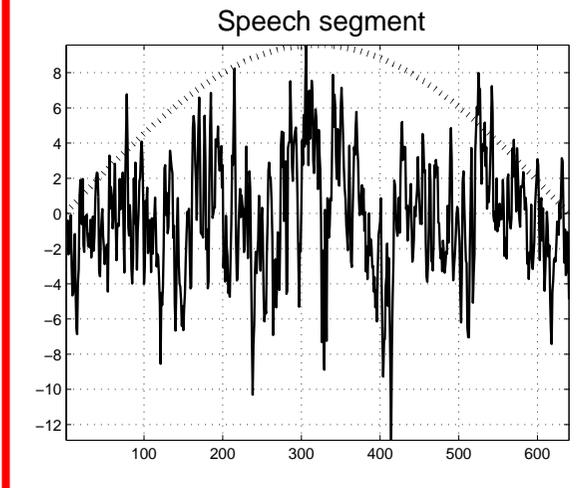
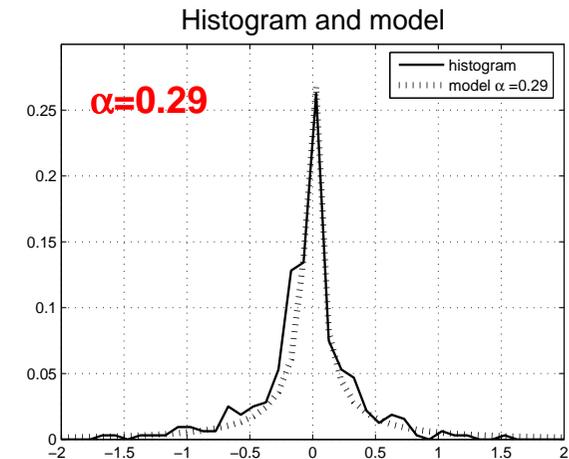
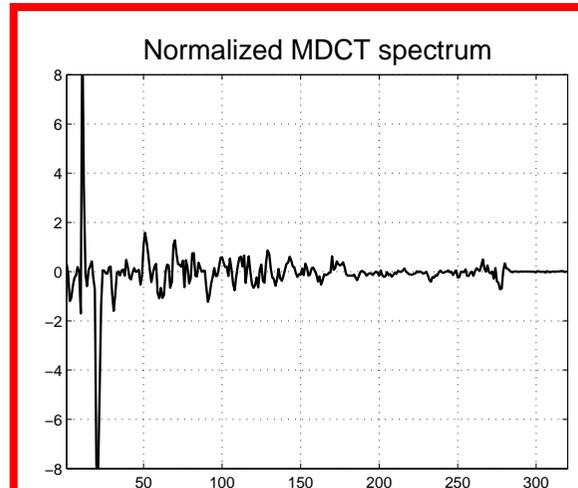
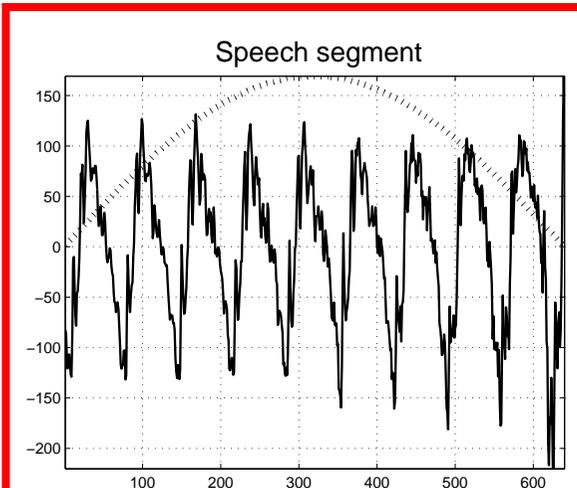
$$\text{So} \quad \hat{\alpha} = F^{-1}\left(\frac{\hat{m}_1}{\sqrt{\hat{m}_2}}\right) = F^{-1}\left(\frac{\sum_{i=1}^n z_i^2}{\sqrt{\sum_{i=1}^n |z_i|}}\right)$$

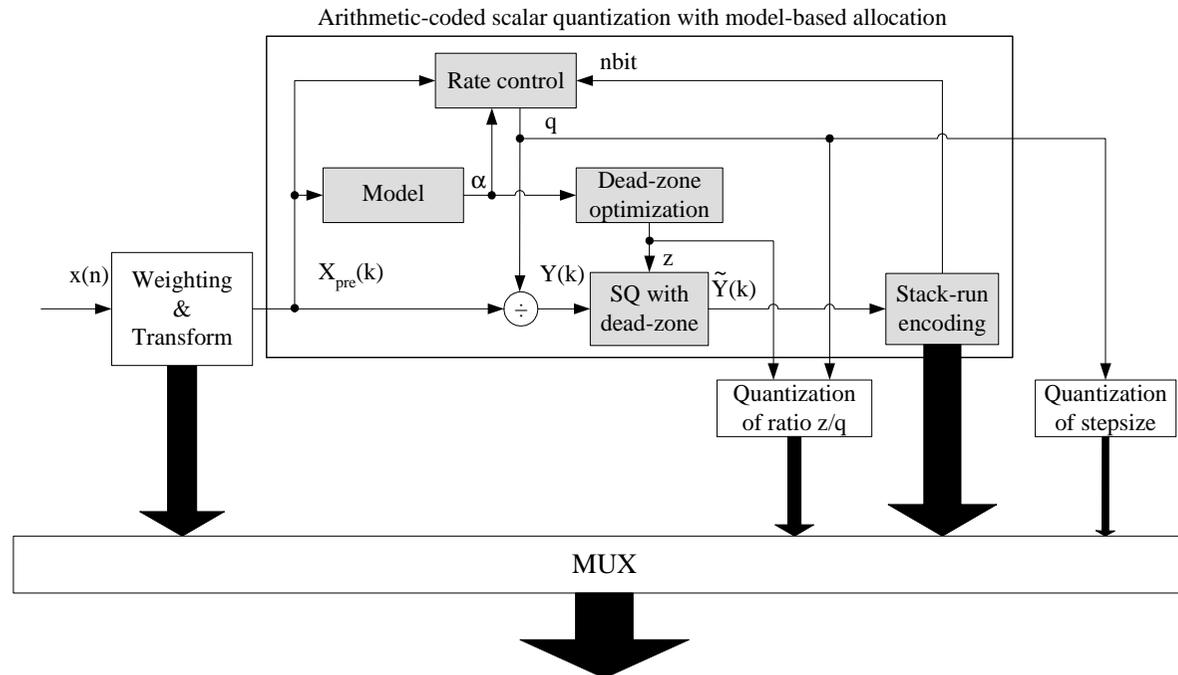
# FlexCode Estimation example: voiced & unvoiced speech



signal segment (time)

spectrum (frequency)

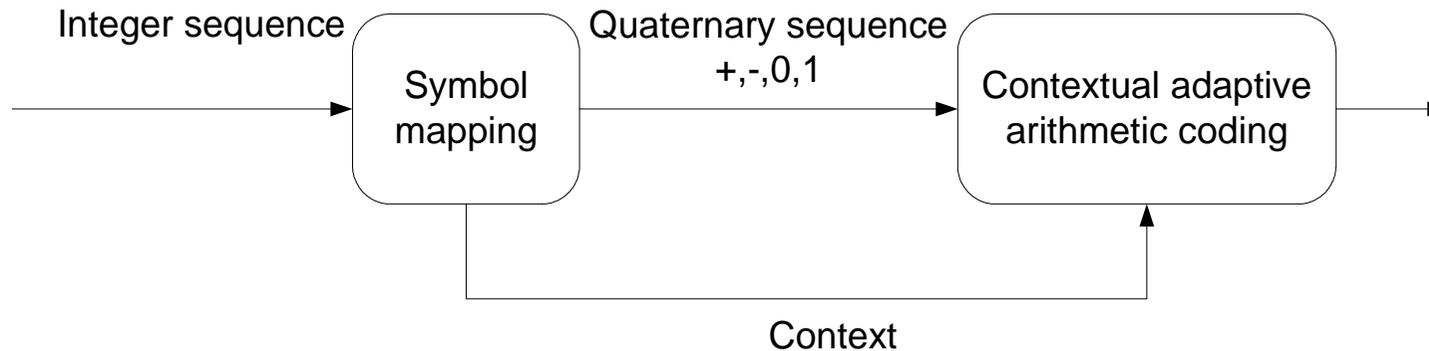




- Input/output signals sampled at 16 kHz
- Frame length of 20 ms with a lookahead of 25 ms (5 ms for LPC analysis and 20 ms for MDCT )
- Effective bandwidth: 50-7000 Hz
- The perceptual weighting filter is defined as:

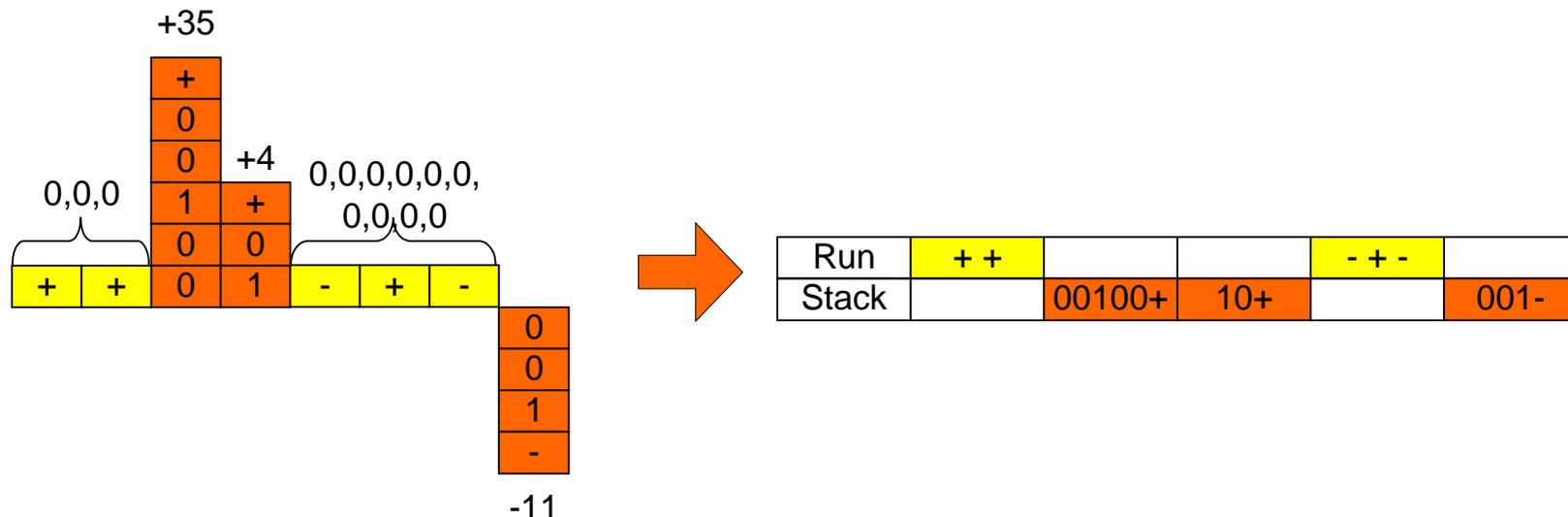
$$W(z) = \frac{A(z/\gamma)}{1 - \beta z^{-1}} \quad \text{with} \quad \beta = 0.75 \quad \text{and} \quad \gamma = 0.92$$

- LPC coefficients quantized with a method based on GMM [Subramaniam 03] [Oger 06]
- MDCT implemented using the fast algorithm of [Duhamel 91]



- Stack-run coding is a **lossless coding method** representing integer sequences
  - Developed for wavelet image coding
- Adaptive arithmetic coding [Witten 87] using a **quaternary alphabet (0, 1, -, +)** and two contexts (one for "**runs**" and another for "**stacks**")
  - A run is a sequence of zeros
  - A stack is a non-zero signed integer

- Mapping rules for stack
  - The binary representation is unsigned
  - MSB is replaced by "+" if the coefficient is positive and "-" if it is negative.
  - The **absolute value is incremented by one**
  - The binary representation of "+4" is "+01" instead of "+00"
- The meanings of the symbol alphabet
  - "0" is used to signify a bit value of 0 in encoding of stack
  - "1" is used for bit value of 1 in stack, but it is not used for the MSB
  - "+" is used to represent the positive MSB of stack and for a bit value of **1 in representing run lengths**
  - "-" is used to represent the negative MSB of stack and for a bit value of **0 in representing run lengths**
- Mapping example for the sequence [0 0 0 +35 +4 0 0 0 0 0 0 0 0 0 0 -11]



- Encoding of **N zero-mean independent variables  $x_i$  of variances  $\sigma_i^2$**
- In case of **high-resolution** the mean square error D [Gersho & Gray 93] is given by

$$D \approx \sum_{i=1}^N h_i \sigma_i^2 2^{-2b_i}$$

where  **$h_i$  is a function of the pdf of the variable  $x_i$**  and  $b_i$  is the number of bits per sample used to code  $x_i$

- For generalized Gaussian variables  $x_i$  the factor  $h_i$  is given by [Parisot 03] :

$$h_i = \frac{\Gamma(1/\alpha_i)^3}{3\alpha_i^2 \Gamma(3/\alpha_i)} e^{2/\alpha_i}$$

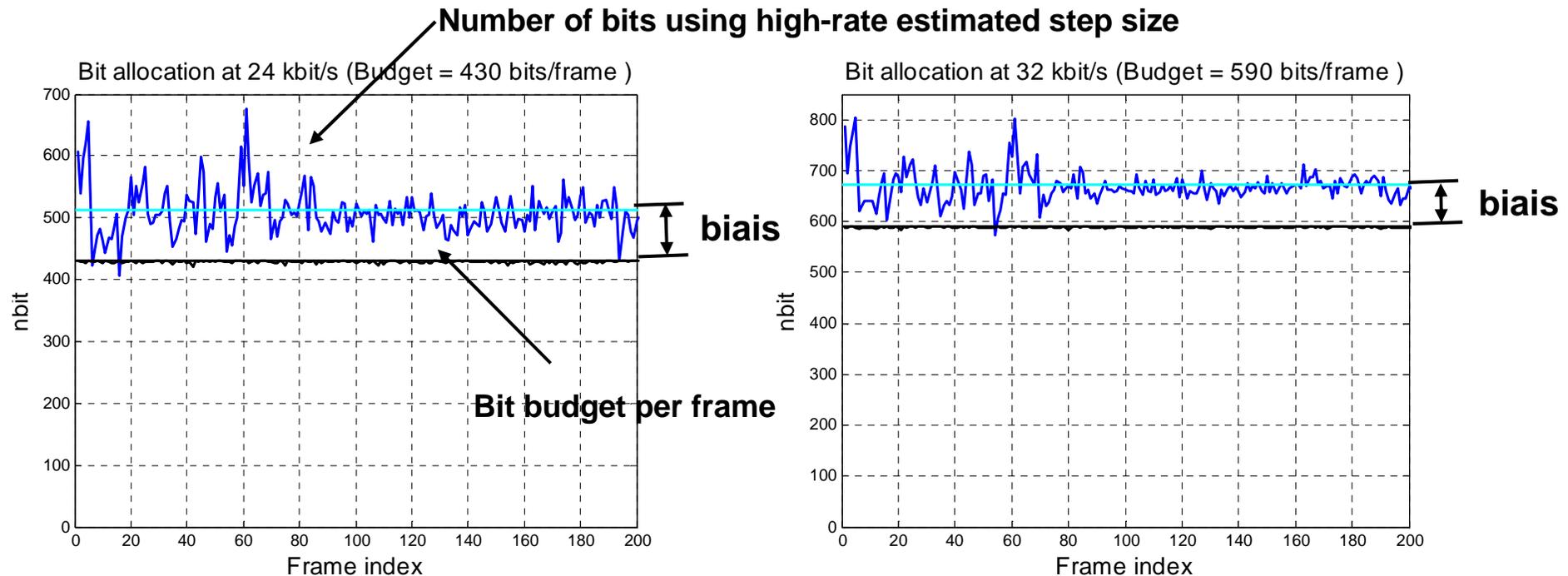
- Encoding of **N zero-mean variables  $x_i$  of variances  $\sigma_i^2$**
- The distortion D can be minimized by **Lagrangian techniques:**

$$J(b_i, \lambda) = D - \lambda \left( \sum_{i=1}^N b_i - B \right) \longrightarrow \lambda_{\text{opt}} = 2 \ln(2) \sum_{i=1}^N h_i \sigma_i^2 2^{-2b_i}$$

where B is the target bit rate

- Hence :
- In **case of high-resolution scalar uniform quantization** with step size q

$$D_{\text{opt}} = \frac{q_{\text{opt}}^2}{12} \longrightarrow q_{\text{opt}} = \sqrt{\frac{6\lambda_{\text{opt}}}{\ln 2}}$$



- Biais due to mismatch high-rate assumption and use of context-based lossless coding instead of zero-entropy coding



A bisection search is used in order to be within the bit budget constraint

- Encoding of **N zero-mean generalized Gaussian variables  $x_i$  of variances  $\sigma_i^2$**
- The distortion D is given by:

$$D(\alpha, z, q) = \frac{1}{\sigma^2} \int_{-z/2}^{z/2} x^2 p_{\sigma, \alpha}(x) dx + \frac{2}{\sigma^2} \sum_{m=1}^{+\infty} \int_{-z/2+(m-1)q}^{z/2+mq} (x - \hat{x}_m)^2 p_{\sigma, \alpha}(x) dx$$

- If the reconstruction level is set to centroid the distortion D is:

$$D(\alpha, z, q) = 1 - \sum_{m=1}^{+\infty} \frac{f_{1,m} \left( \alpha, \frac{z}{\sigma}, \frac{q}{\sigma} \right)^2}{f_{0,m} \left( \alpha, \frac{z}{\sigma}, \frac{q}{\sigma} \right)} \quad \text{where} \quad f_{n,m} \left( \alpha, \frac{z}{\sigma}, \frac{q}{\sigma} \right) = \int_{z/2\sigma+(m-1)q/\sigma}^{z/2\sigma+mq/\sigma} x^n p_{1,\alpha}(x) dx$$

- If the reconstruction level is set to mid-value the distortion D is:

$$D(\alpha, z, q) = 1 + 2 \sum_{m=1}^{+\infty} \left( \frac{1}{2} \frac{z}{\sigma} + \left( m - \frac{1}{2} \right) \frac{q}{\sigma} \right)^2 f_{0,m} \left( \alpha, \frac{z}{\sigma}, \frac{q}{\sigma} \right) - 4 \sum_{m=1}^{+\infty} \left( \frac{1}{2} \frac{z}{\sigma} + \left( m - \frac{1}{2} \right) \frac{q}{\sigma} \right) f_{1,m} \left( \alpha, \frac{z}{\sigma}, \frac{q}{\sigma} \right)$$

- The bit rate  $R$  is given by:

$$R = -P(0) \log_2 P(0) - 2 \sum_{m=1}^{+\infty} P(m) \log_2 P(m)$$

- With  $P(m) = \int_{z/2+(m-1)q}^{z/2+mq} x^n p_\alpha(x) dx = f_{0,m} \left( \alpha, \frac{z}{\sigma}, \frac{q}{\sigma} \right)$

- So the bit rate  $R$  is given by:

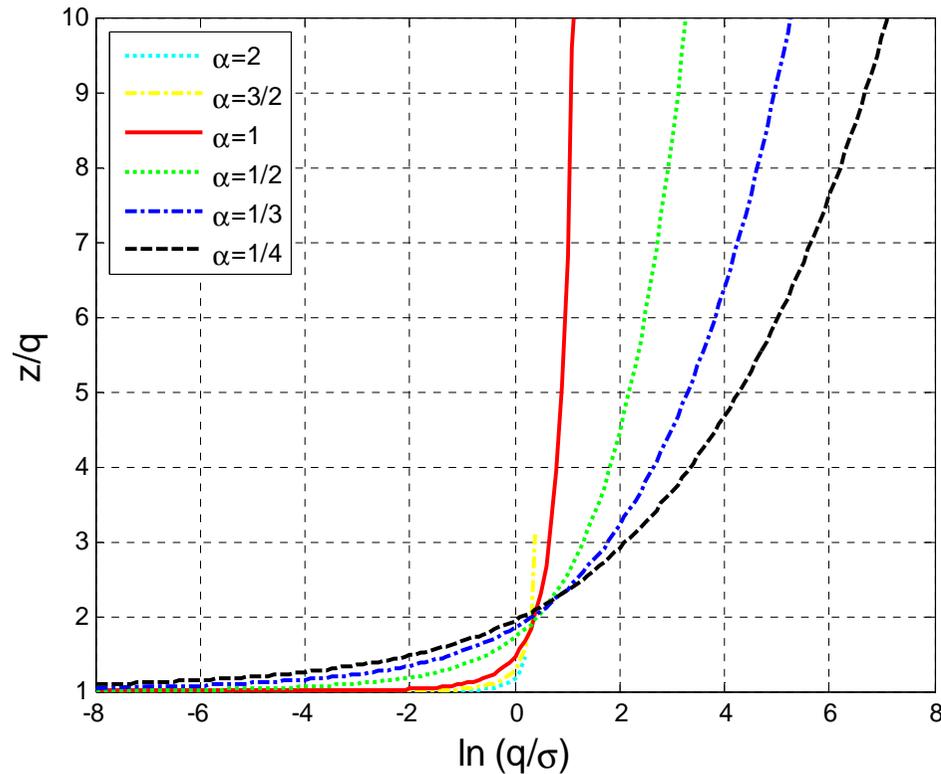
$$R = -f_{0,0} \left( \alpha, \frac{z}{\sigma} \right) \log_2 f_{0,0} \left( \alpha, \frac{z}{\sigma} \right) - 2 \sum_{m=1}^{+\infty} f_{0,m} \left( \alpha, \frac{z}{\sigma}, \frac{q}{\sigma} \right) \log_2 f_{0,m} \left( \alpha, \frac{z}{\sigma}, \frac{q}{\sigma} \right)$$

- Encoding of **N zero-mean generalized Gaussian variables  $x_i$  of variances  $\sigma_i^2$**
- The distortion D can be minimized by **Lagrangian techniques:**

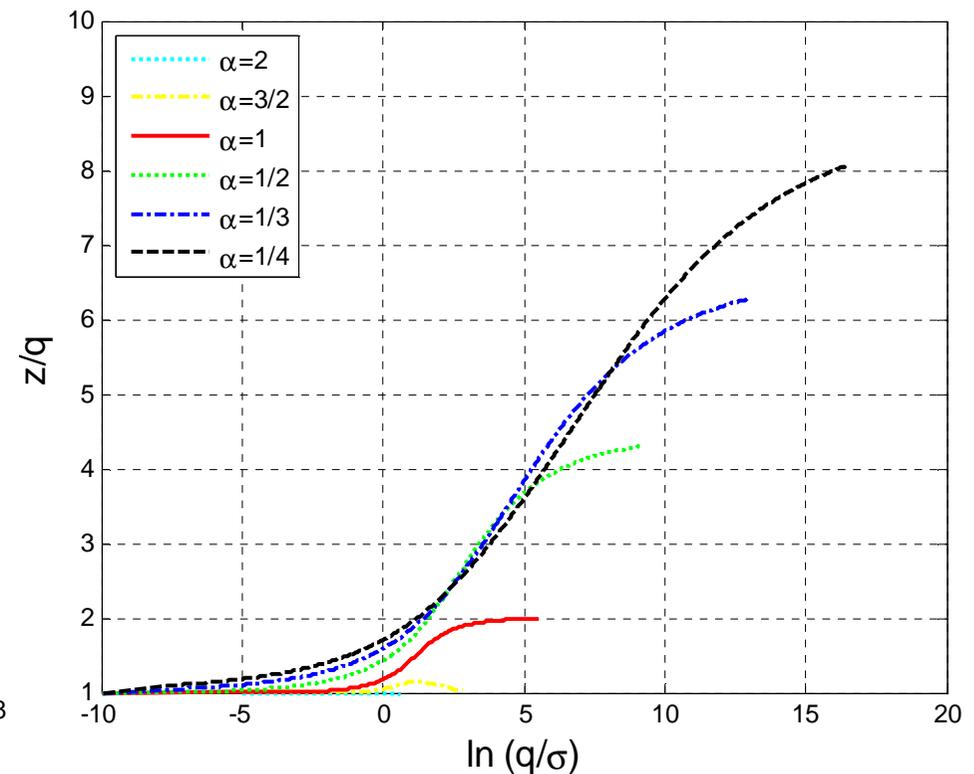
$$J(z_i, q_i, \lambda) = \sum_{i=1}^N \sigma_i^2 D\left(\alpha_i, \frac{z_i}{\sigma_i}, \frac{q_i}{\sigma_i}\right) + \lambda \left( \sum_{i=1}^N a_i R\left(\alpha_i, \frac{z_i}{\sigma_i}, \frac{q_i}{\sigma_i}\right) - R_{\text{target}} \right)$$

$$\begin{array}{l}
 \left. \begin{array}{l}
 \frac{\partial D}{\partial \tilde{z}}(\alpha_i, \tilde{z}_i, \tilde{q}_i) = \frac{\partial D}{\partial \tilde{q}}(\alpha_i, \tilde{z}_i, \tilde{q}_i) \\
 \frac{\partial R}{\partial \tilde{z}}(\alpha_i, \tilde{z}_i, \tilde{q}_i) = \frac{\partial R}{\partial \tilde{q}}(\alpha_i, \tilde{z}_i, \tilde{q}_i)
 \end{array} \right\} \longrightarrow \text{Optimization of the deadzone } z \\
 \\
 \left. \begin{array}{l}
 \frac{\partial D}{\partial \tilde{q}}(\alpha_i, \tilde{z}_i, \tilde{q}_i) = -\frac{\lambda a_i}{\sigma_i^2} \\
 \frac{\partial R}{\partial \tilde{q}}(\alpha_i, \tilde{z}_i, \tilde{q}_i) = \frac{\lambda a_i}{\sigma_i^2}
 \end{array} \right\} \quad \sum_{i=1}^N a_i R(\alpha_i, \tilde{z}_i, \tilde{q}_i) = R_{\text{target}}
 \end{array}$$

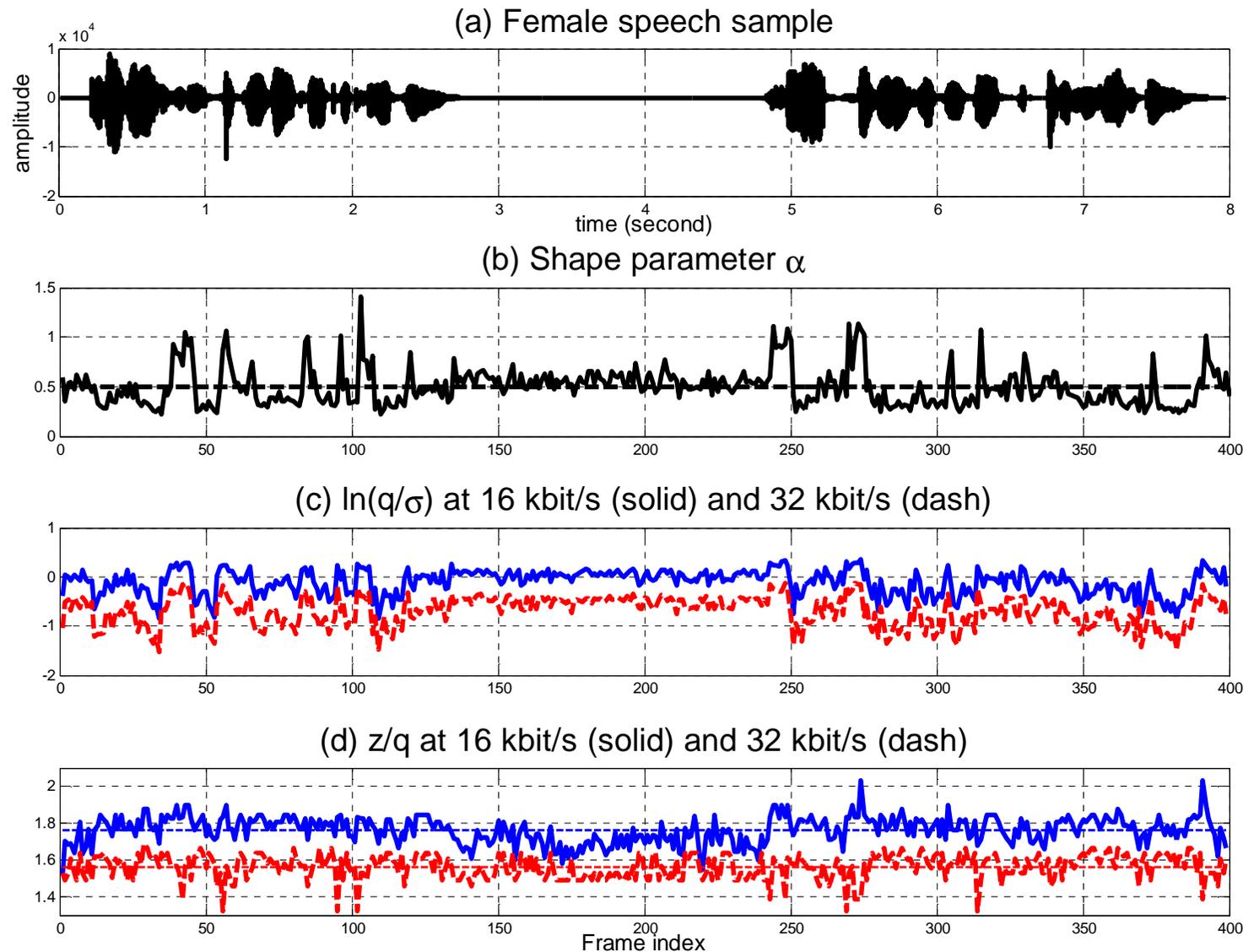
where  $\alpha_i$ ,  $z_i$ , and  $q_i$  are respectively the shape parameter, the deadzone and the stepsize



Scalar quantizer with reconstruction levels set to mid-value



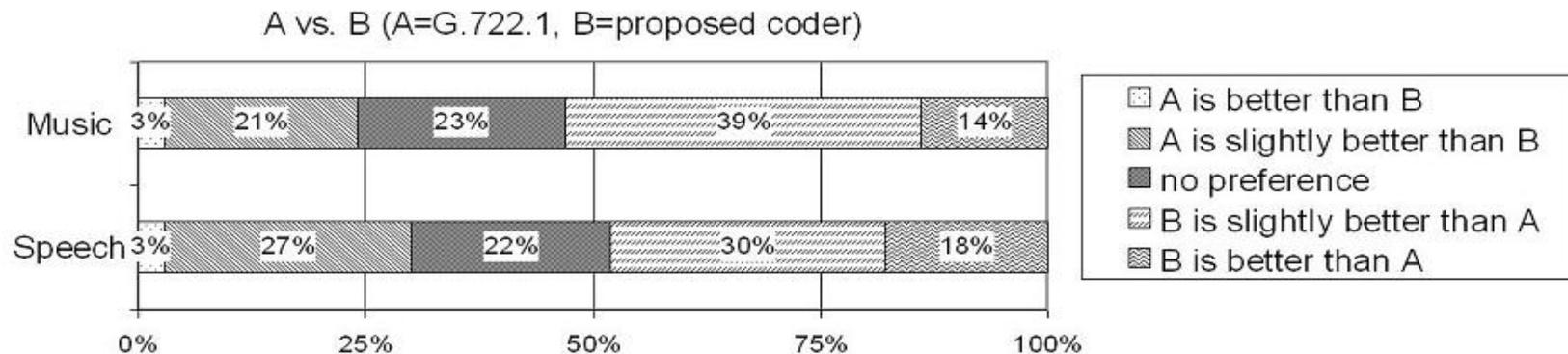
Scalar quantizer with reconstruction levels set to optimal centroid (Lloyd-Max)



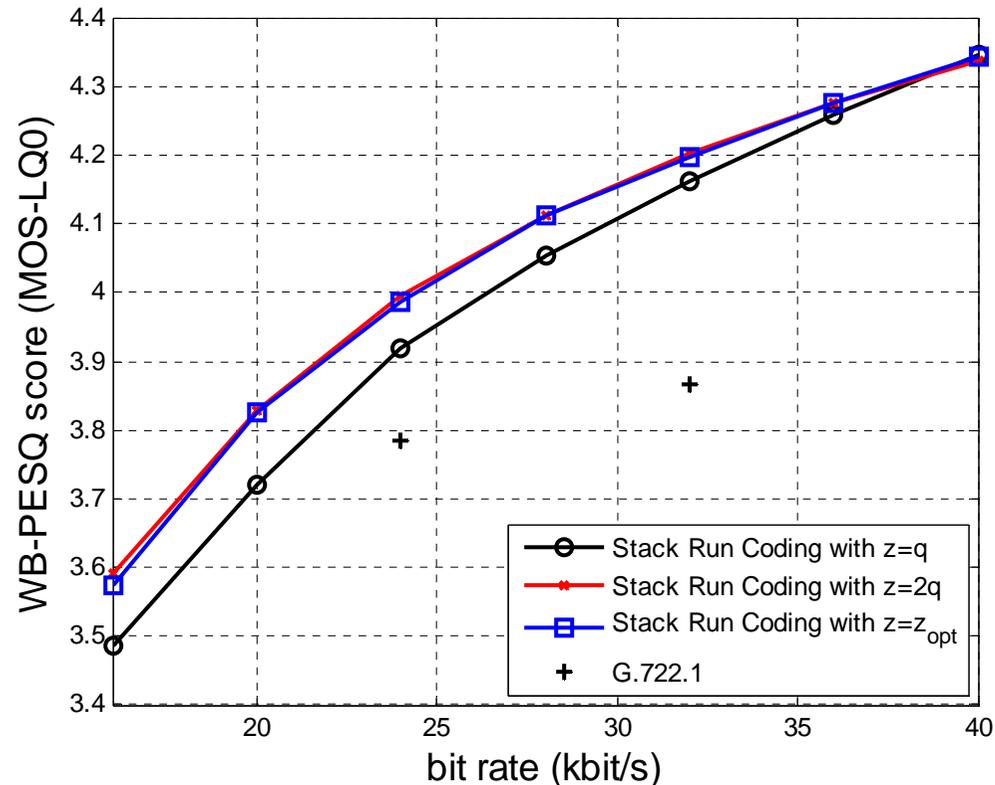
# FlexCode Subjective quality results (no deadzone)



- 16 clean music samples (4 types  $\times$  4 sentence-pairs) of 8 seconds
- 24 clean speech samples in French language (6 male and female speakers  $\times$  4 sentence-pairs) of 8 seconds
- Two AB test at 24 kbit/s : one for speech, another for music

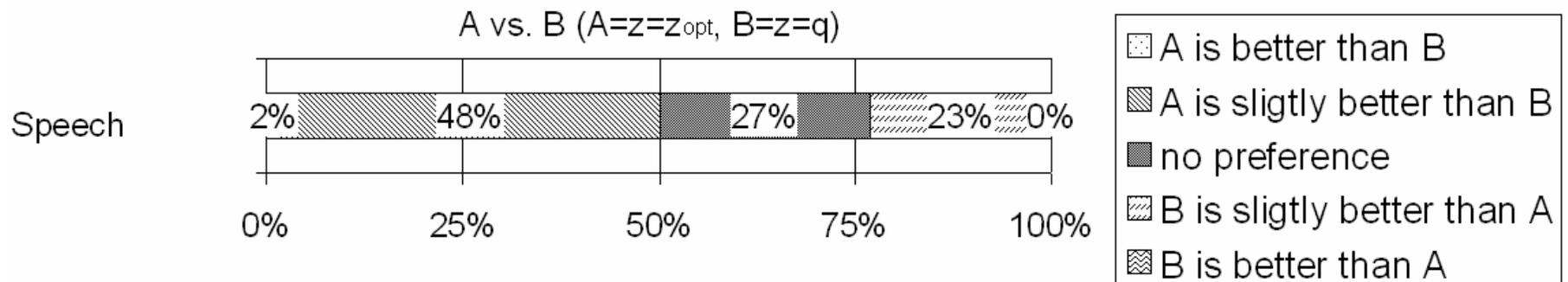


- 8 expert listeners
- The stack-run coding (z=q) was preferred:
  - In 53% cases for music
  - In 48% cases for speech
- Informal listening tests at 32 kbit/s
- **Stack-run coding (z=q) is better than ITU-T G.722.1 at 24 kbit/s and equivalent at 32 kbit/s**



- 24 clean speech samples in French language (6 male and female speakers  $\times$  4 sentence-pairs) of 8 s
- Proposed coder: predictive MDCT coder with stack-run coding
- Results are presented with noise injection (injection similar to 3GPP AMR-WB+)
- These **objective results suggest that the inclusion of a dead-zone improves the performance**

- 20 clean speech samples in French language (5 male and female speakers  $\times$  4 sentence-pairs) of 8 seconds

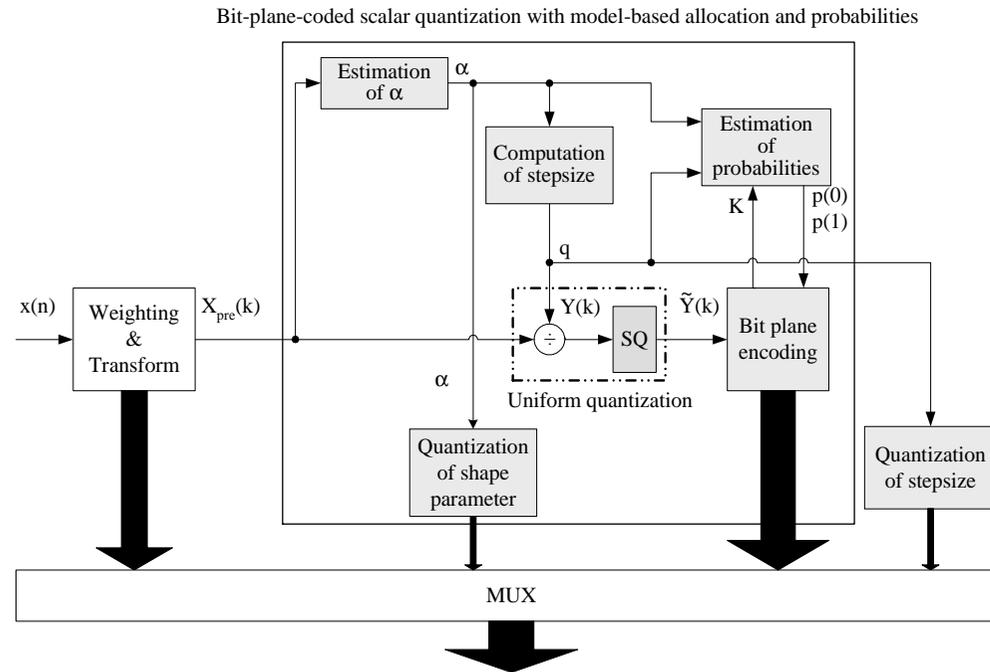


- One AB test at 24 kbit/s for speech:
  - 9 expert listeners
  - Stack-run coding with  $z=z_{opt}$  was preferred in 50% cases for speech
- Informal listening tests at 32 kbit/s
- Stack-run coding with  $z=z_{opt}$  is better than stack-run coding with  $z=q$  at low bitrate and is equivalent at high bitrate.**

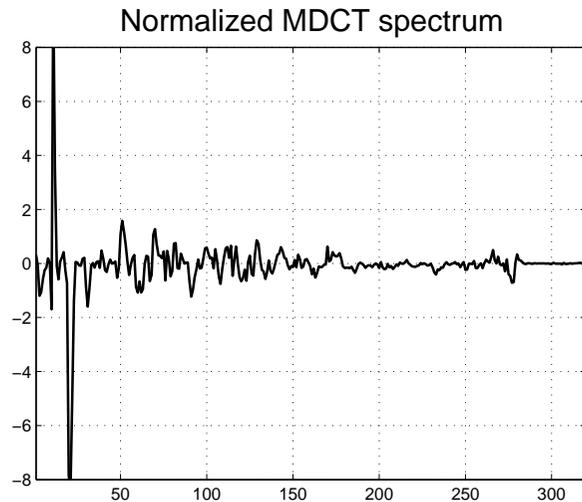
- Algorithmic delay:
  - 45 ms (20 ms for the frame, 20ms for the MDCT and 5 ms for the lookahead) for the stack-run coder
  - 40 ms for the ITU-T G.722.1
- The computational complexity of ITU-T G.722.1 is very low
- The computational complexity of the stack-run coder is higher due to the use of bisection search for bit rate matching
  - Stack-run coding is performed several times per frame
- Storage requirements for the stack-run coder are low
  - Parameters for GMM based LPC quantization
  - MDCT tables (can be computed on lines)

Comparison between stack-run coding and ITU-T G.722.1

	Stack-run coding with $z=q$	Stack-run coding with $z=z_{opt}$	ITU-T G.722.1
Music at 24 kbit/s			
Music at 32 kbit/s			
Speech at 24 kbit/s			
Speech at 32 kbit/s			

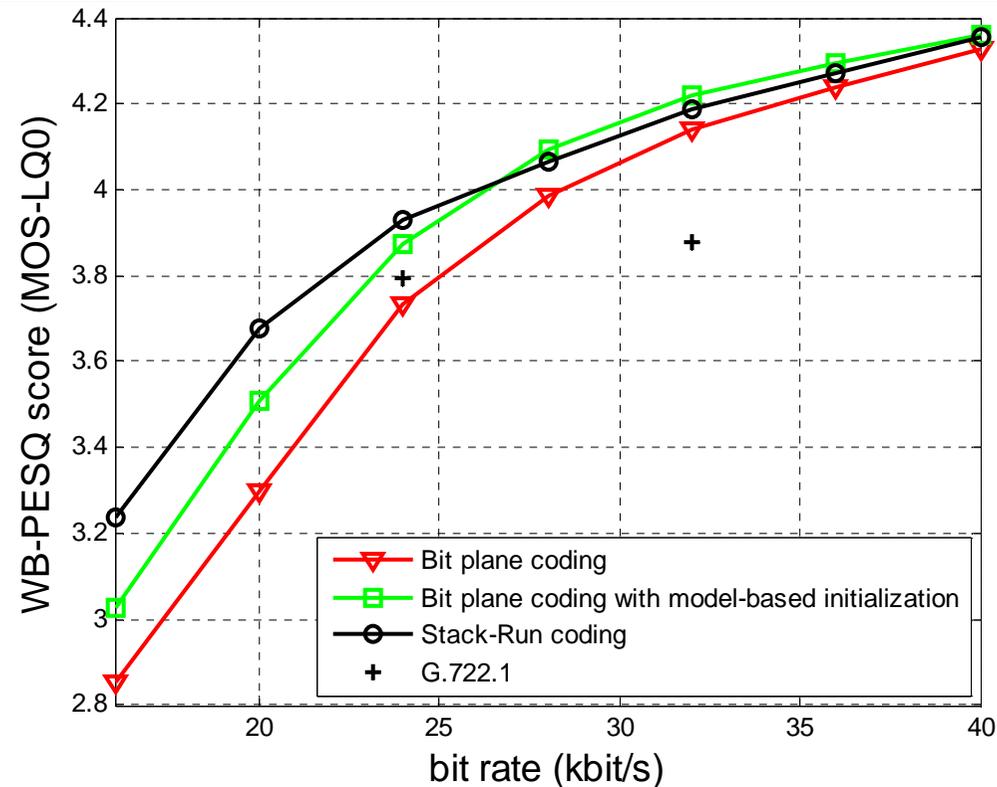


- Principle: replace stack-run coding by bit plane coding.
  - Implicit rate control
- ➔ Computational complexity is much more lower than stack-run coding



$Y =$	-2	+5	-3	.....	+1	0	+1	
	1	0	1	.....	0	0	0	→ Bit plane of signs
$\uparrow$								
$2^2$	0	1	0	0.....0	0	0	0	→ $P_2 = \text{MSB}$
$\uparrow$								
$2^1$	1	0	1	0.....0	0	0	0	
$\uparrow$								
$2^0$	0	1	1	0.....1	1	0	1	→ $P_0 = \text{LSB}$
$\downarrow$								

- The normalized MDCT spectrum  $X_{\text{pre}}(k)$  is scalar quantized and we get an integer sequence  $\tilde{Y}(k)$ .
- This integer sequence is decomposed in binary format.
- The symbol probabilities in bit planes are estimated on the model of the pdf of  $X_{\text{pre}}(k)$



- 24 clean speech samples in French language (6 male and female speakers  $\times$  4 sentence-pairs) of 8 s
- Proposed coder: predictive MDCT coder with bit-plane coding
- Results are presented without noise injection
- These **objective results suggest that the speech quality of the proposed coder with model-based initialization of symbol probabilities is equivalent to reference coders at high bitrate**

- We proposed a predictive MDCT coder with generalized Gaussian modeling for wideband speech and audio signals
- Generalized Gaussian modeling is used to:
  - Estimate the optimal step size
  - Optimize the deadzone
  - Estimate symbol probabilities in bit planes
- Next step: Include sensitivity matrix into model-based coder
  - Linear-predictive filter → signal-adaptive transform

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Thank you!