ARTIFICIAL BANDWIDTH EXTENSION USING THE CONSTANT Q TRANSFORM

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Introduction

- public telephone networks limit the bandwidth of speech signals to 300-3400Hz
- intelligibility for unvoiced phonemes is generally lower than that for voiced phonemes because their spectra extends beyond 3400Hz
- wider bandwidths generally correspond to higher quality speech [1]
- artificial bandwidth extension (ABE) methods estimate missing frequency components to compensate for the consequential loss speech quality and intelligibility
- most ABE algorithms are based either on the classical source-filter model OR employ short time Fourier transform (STFT) for spectral analysis
- the STFT offers a fixed frequency resolution, and is equivalent to a bank of filters with variable Q factors
- however, the human auditory system exhibits constant Q characteristics between 500Hz to 20kHz

Contributions

- Application of the constant Q transform (CQT), a more perceptually motivated approach to spectral analysis, to ABE.

The constant Q transform (CQT)

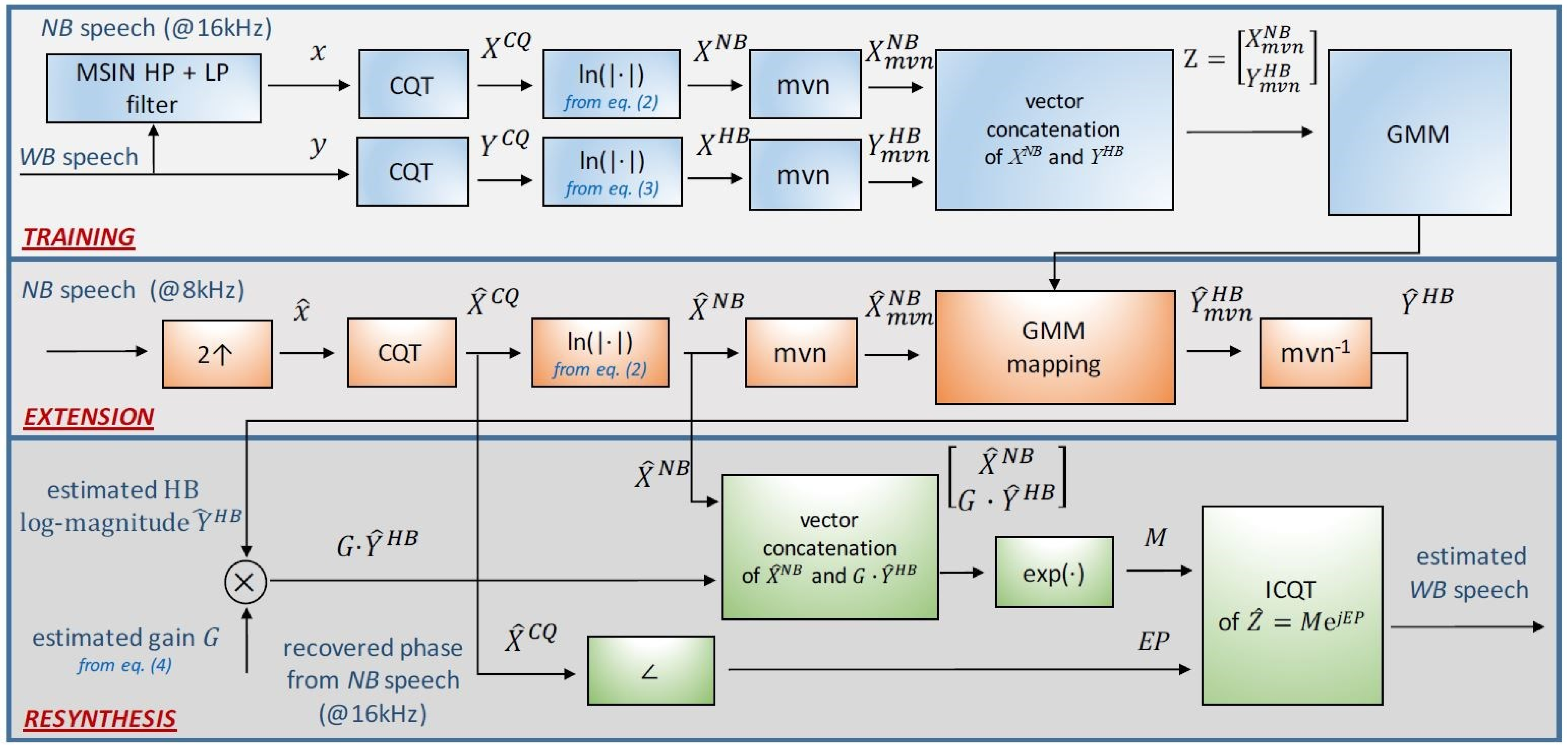
- Uncertainty principle: time and frequency content cannot be measured precisely at the same time
- Q factor is defined as $Q = \frac{f_k}{\delta f}$
- the bandwidth of each STFT filter is constant, whereas the Q factor increases from low to high frequencies
- however, human perception approximates a constant Q transform between 500 Hz and 20 kHz
- CQT: introduced by Youngberg and Boll [2], and refined over the years [3];

$$X^{CQ}(k) = \frac{1}{N_k}\sum_{n<N_k} x(n) w_{N_k}(n) e^{-2\pi i n \frac{f_k}{f_s}}$$

$$N_k = Q\frac{f_s}{f_k} \qquad Q = (2^{\frac{1}{B}} - 1)^{-1}$$

where B is the number of bins per octave

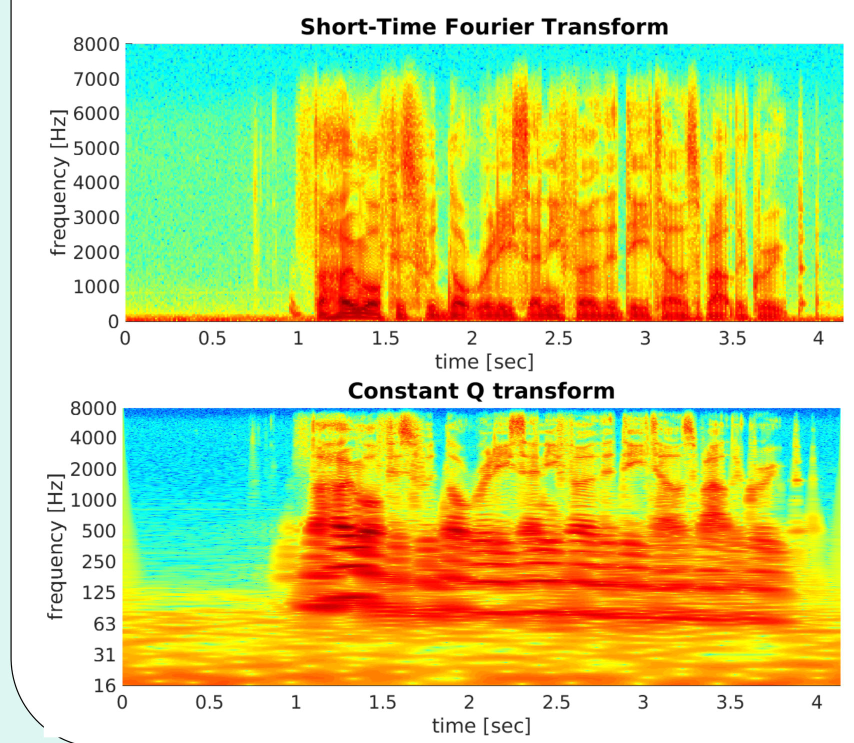
- filter center frequencies are geometrically spaced
- greater frequency resolution for lower frequencies and a greater time resolution for higher frequencies



Block diagram of the CQT-based ABE system.

Experimental Setup

- Database: TSP speech database [4] consisting of 1378 utterances spoken by 12 male and 12 female speakers.
- CQT Parameters: B=48 bins/octave, f_{max} = 8000 Hz, f_1 = 250 Hz
- Mapping: GMM regression (using 512 components)
- input NB (250-3.4kHz) features – 187D, Output HB (3.4-8kHz) features- 52D
- during resynthesis, gain ***G*** corrects the energy of estimated HB which is learned through polynomial regression of order 4.
- whereas phase is copied from HB of upsampled NB CQT signal



Spectrograms of the utterance 'the woman is a star who has grown to love the limelight' for a male speaker in the ASVspoof database. Spectrograms computed with the short-time Fourier Transform (top) and with the constant Q transform (bottom)

Experimental Results

Gain	Phase	Train Mean (σ)	Test Mean (σ)
-	OP	3.01 (0.72)	5.28 (1.51)
-	EP	3.21 (0.71)	5.39 (1.49)
OG	OP	1.89 (0.37)	3.13 (0.67)
OG	EP	2.16 (0.38)	3.30 (0.67)
EG	OP	2.46 (0.40)	4.64 (1.06)
EG	**EP**	2.66 (0.42)	4.77 (1.05)

***RMS-LSD** results (in dB) with and without gain normalization and different phase extensions. OG - oracle gain, EG – estimated gain, OP - oracle phase, EP - estimated phase. EG-EP is the proposed method.*

Comparison B → A	MOS
EG-EP → NB	1.12
OG-EP → NB	1.14
EG-EP → WB	-1.42
OG-EP → WB	-1.03

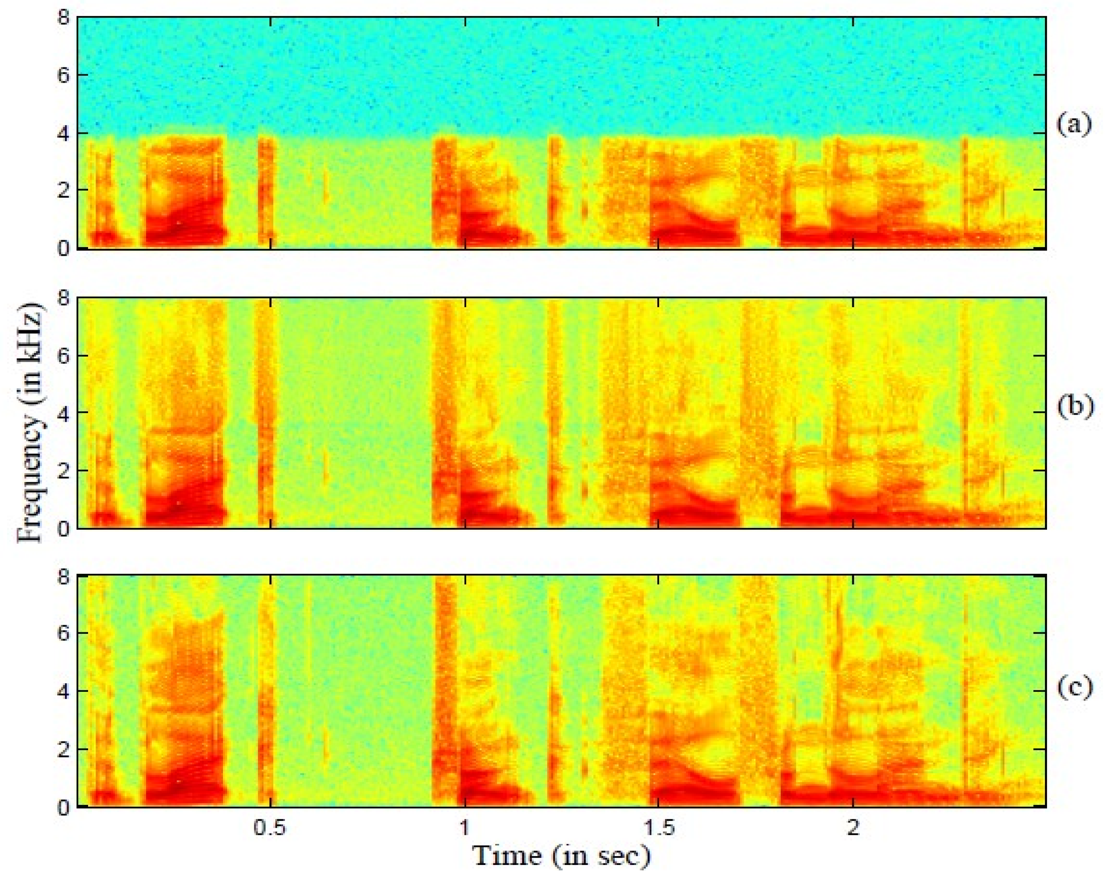
*Comparison based **MOS** for EP with EG and OG. EG-EP is the proposed method (files used for the subjective evaluation are available at **http://audio.eurecom.fr/content/media**)*

Conclusions and Future work

- the CQT is a perceptually motivated approach to time-frequency analysis
- ABE using the CQT produces higher quality, higher bandwidth speech signals using the CQT
- the accurate estimation of the spectral magnitude and gain is critical

Future Work

- analysis and optimisation of the CQT over traditional STFT
- future work will investigate the application of ABE to music signals



Spectrograms of an upsampled NB speech (a), artificially extended WB speech (b) and original WB speech (c).

Acknowledgements

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Selected References

[1] P. Jax and P. Vary, "On artificial bandwidth extension of telephone speech", *Signal Processing*, vol .83 , no. 8, pp. 1707-1719, 2003

[2] J. Youngberg and S. Boll, "Constant-q signal analysis and synthesis," in *Proc. of ICASSP*,1978.

[3] J. Brown, "Calculation of a constant Q spectral transform," *Journal of the Acoustical Society of America*, vol. 89, no. 1, pp. 425–434, January 1991.

[4] P. Kabal, "TSP Speech Database", McGill University. [Online] Available: http://www-mmsp.ece.mcgill.ca/Documents/Data/.

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