MODELLING SPEECH QUALITY FOR NARROWBAND AND WIDEBAND SILK CODEC FOR VOIP APPLICATIONS

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Overview

- Introduction
- SILK Codec
- Overview of speech quality models
- Testbed setup
- SILK quality modelling based on E-model
- Model’s performance
- Subjective evaluation of the model
- Summary

Introduction

- VoIP networks , Skype
- Factors affect speech quality
- Proprietary software, open-source codec
- How to measure speech quality

SILK Codec

- Speech codec and compression format develo
  - Sup
  - Exp … cordless pho
  - Freely available for third party users

Mode	Fs (Hz)	BR(kbps)
Narrowband	8000	6 – 20
Mediumband	12000	7 – 25
Wideband	16000	8 – 30
Super Wideband	24000	12 – 40

- Highly scalable codec
  - Network bandwidth
  - Send bitrate
  - Complexity

Overview of measurement methods

- Subjective methods
- Objective models
  - Computational models
    - E-model
    - Regression based models
    - Machine learning
  - Signal-based models
    - PESQ (ITU-T P862) and PESQ-WB
    - POLQA (ITU-T P.863)

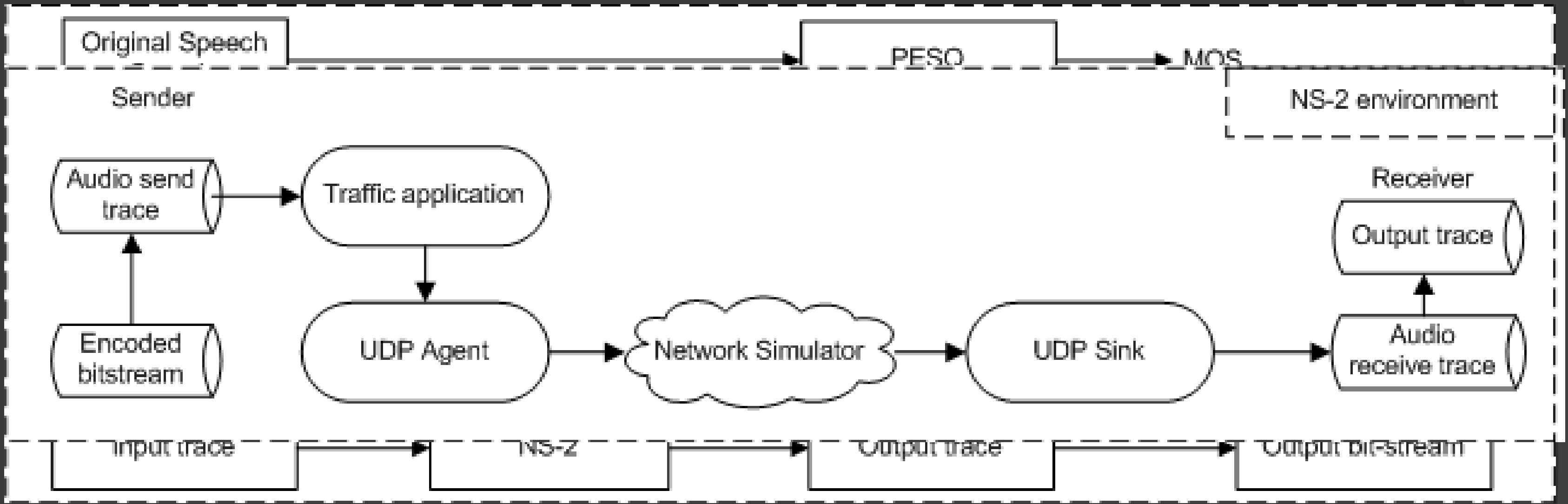
Speech quality measurement models

- Measure the end-to-end quality (MOS)
- Impact of network conditions (i.e., packet loss, delay, etc.)
- Maintain a good service quality during the call
- Application level adaptation
  - Bitrate
  - Sampling frequency
  - Packetization
  - Inband FEC
- Online monitoring and control

Aims

- Low complexity, reliable measurement model to quantify the speech quality for SILK codec
- The model can be used in adaptation, FEC tuning and user satisfaction measurement
- Use both application level and network level parameters to predict the speech quality of VoIP calls.
- Can have direct applications in monitoring and adaptation of VoIP applications

Testbed setup



- Packet loss (0% to 30% with 5% intervals)
- 16 British English speech samples (ITU-T P.50)
  - 8 male and 8 female
- 15 different initial seeds

Modelling based on E-model

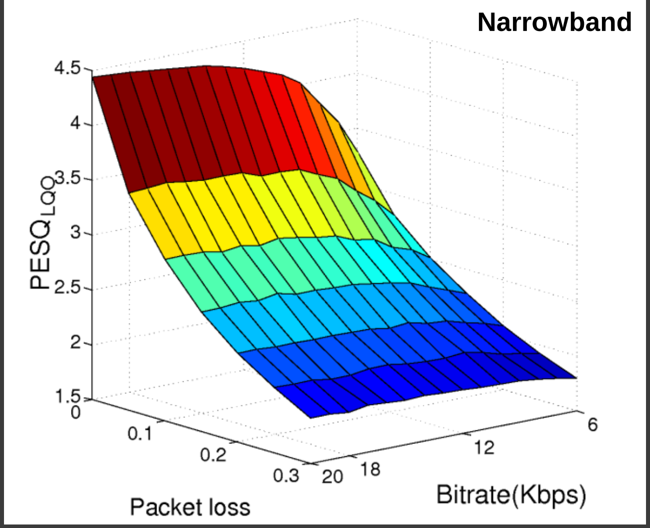
E-model (ITU-T G.107) :

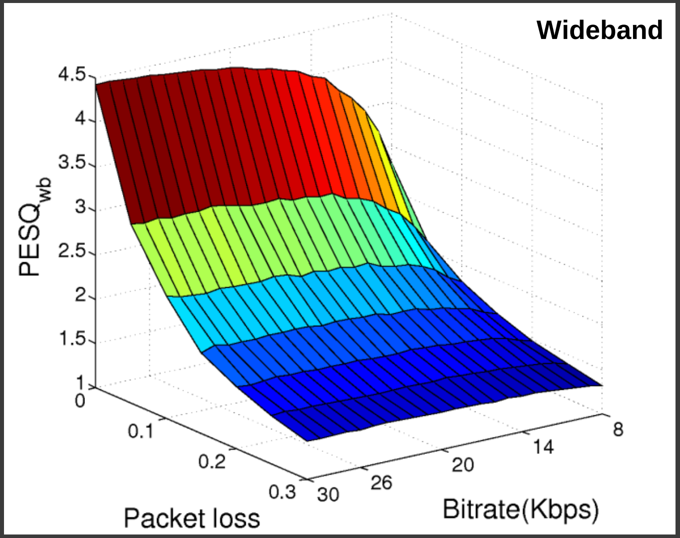
$$R = R_o - Id - Ie + A$$

- R_o : basic signal-to-noise ratio (SNR)
  - 93.2 for NB
  - 129 for WB
- Id : is the impairments caused by delay factors
- Ie : includes distortions by codec and packet loss
- A : Advantage factor.

$$R = R_o - Ie(bitrate, loss)$$

Modelling $Ie.eff$





$$R = 3.026MOS^3 - 25.314MOS^2 + 87.060MOS - 57.336$$

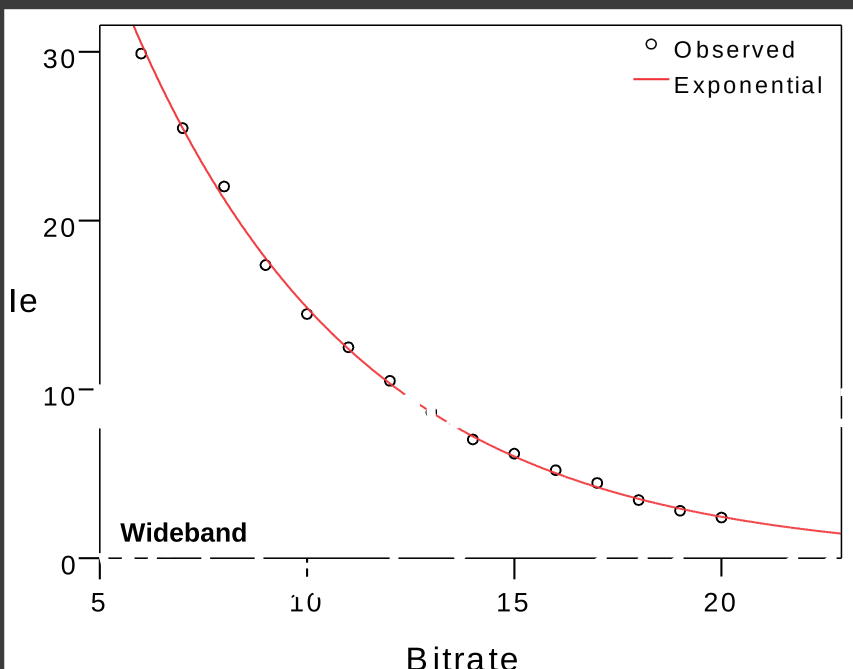
$$Ie.eff = Ro - R$$

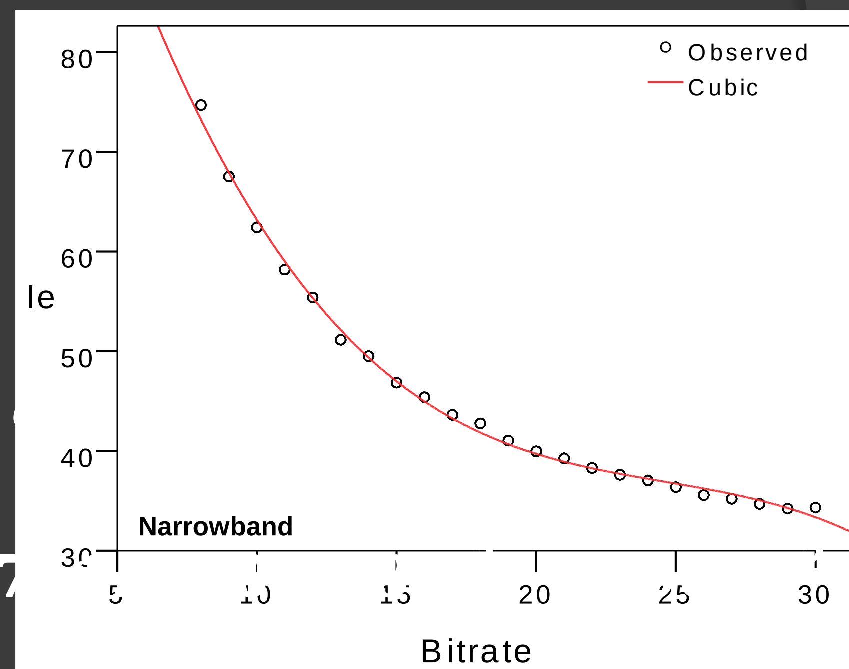
$$\mathrm{I_{e.eff}} = \boldsymbol{I_e} + (\boldsymbol{R_0} - \boldsymbol{I_e}) \frac{\boldsymbol{Ppl}}{\boldsymbol{Ppl} + \boldsymbol{Bpl}}$$

Modelling Ie

Ie values are obtained by taking the $Ie.eff$ values in 0% packet loss.

Each bitrate level, has a fixed Ie value.





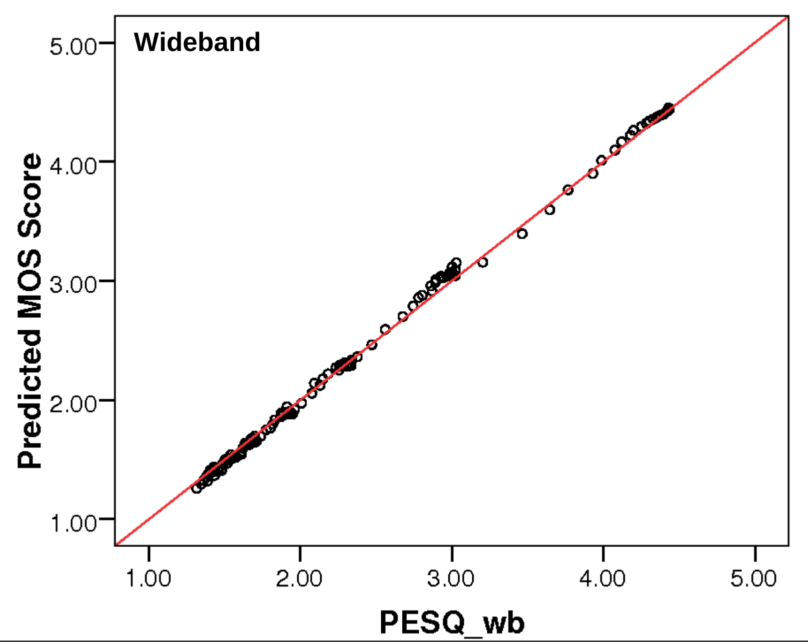
Modelling $\boldsymbol{Bpl}$

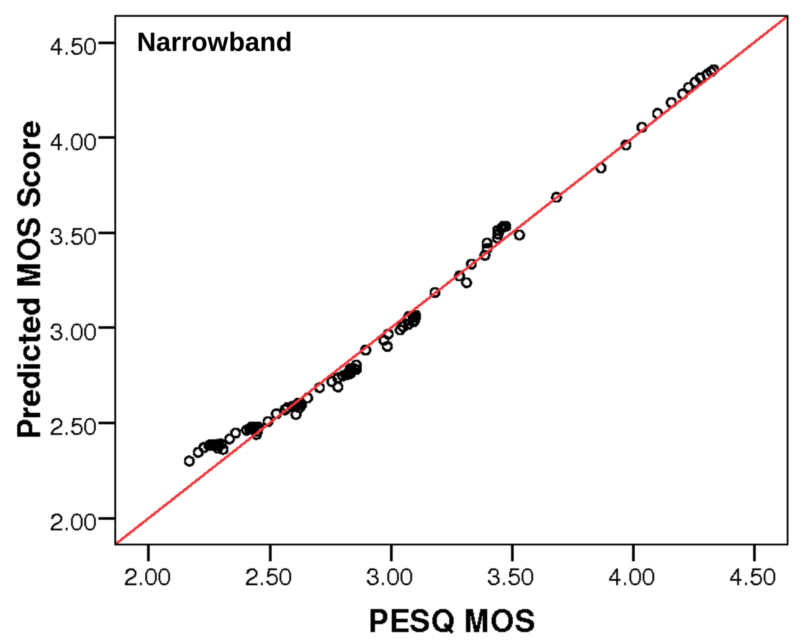
- Non-linear regression analysis of the model with our training set
- $Ie.eff$ values obtained in the previous step

$$\boldsymbol{Bpl = \propto_0 + \propto_1 B + \propto_2 B^2 + \propto_3 B^3}$$

Coeff	NB	WB
$\propto_0$	36.836	122.617
$\propto_1$	−4.893	16.391
$\propto_2$	0.287	0.011
$\propto_3$	−0.006	0
r^2	99.94%	99.8%
RMSE	0.9922	0.9871

Model vs PESQ results





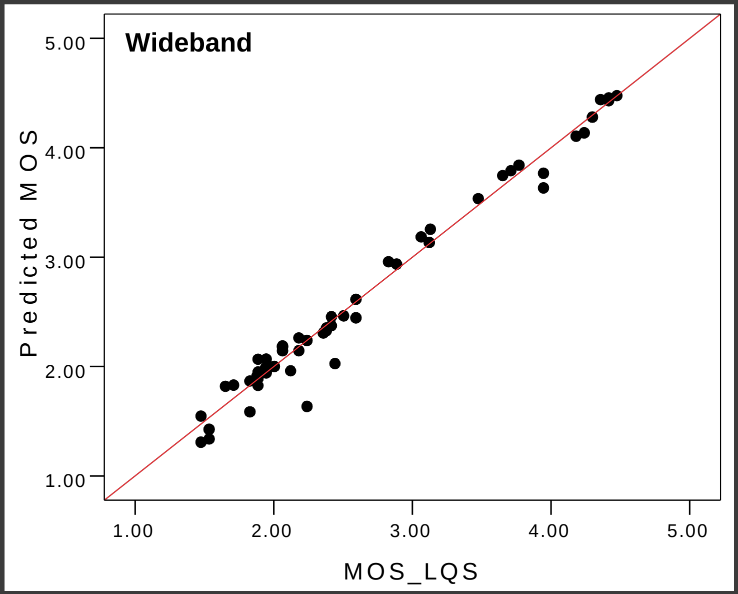
WB mode : 98%

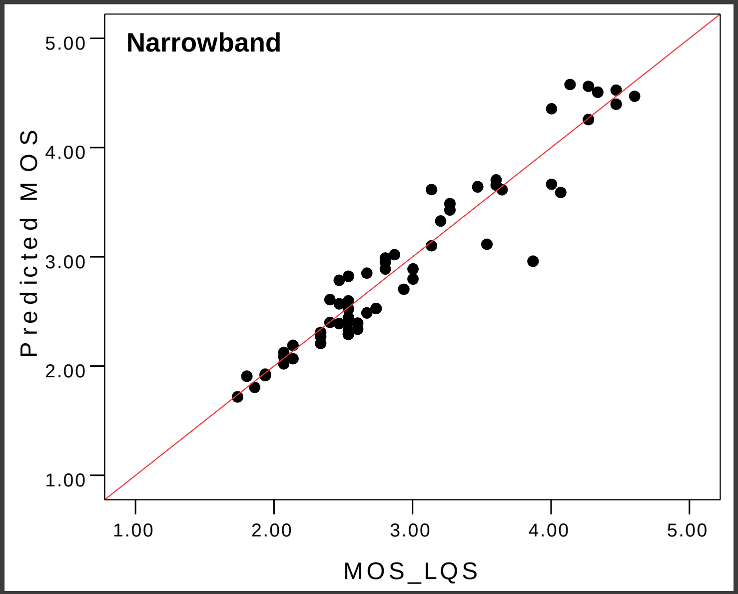
NB mode : 99%

Subjective evaluation of the model

- Two sets of subjective tests were carried out for NB and WB samples according to ITU-T P.800 recommendation.
- Sessions were conducted in 2 separate days
  - 60 Samples per session divided into 4 groups
  - Samples were randomly assigned for each user

Subjective results





Measure	WB	NB
r^2	97.83%	91.86%
Max error	0.59	0.901
Min error	−0.190	−0.486
RMSE	0.1362	0.2272

Conclusion

- Developed a regression-based model, that can measure the speech quality of VoIP calls for Skype's SILK codec for VoIP applications
- The model is a reference-free, regression-based model based on sender bitrate and packet loss parameters
- Validated our developed model with subjective test results
- Correlation of 91% for narrowband and 97% for wideband.
- Can be implemented in mobile devices or softphones
  - Monitoring, control, adaptation and tuning for optimal user satisfaction in real time
- Aspects of this work are generic and can be adopted to other multi-rate codecs

Thank you

- Q & A

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