A machine-learning approach to application of intelligent artificial reverberation

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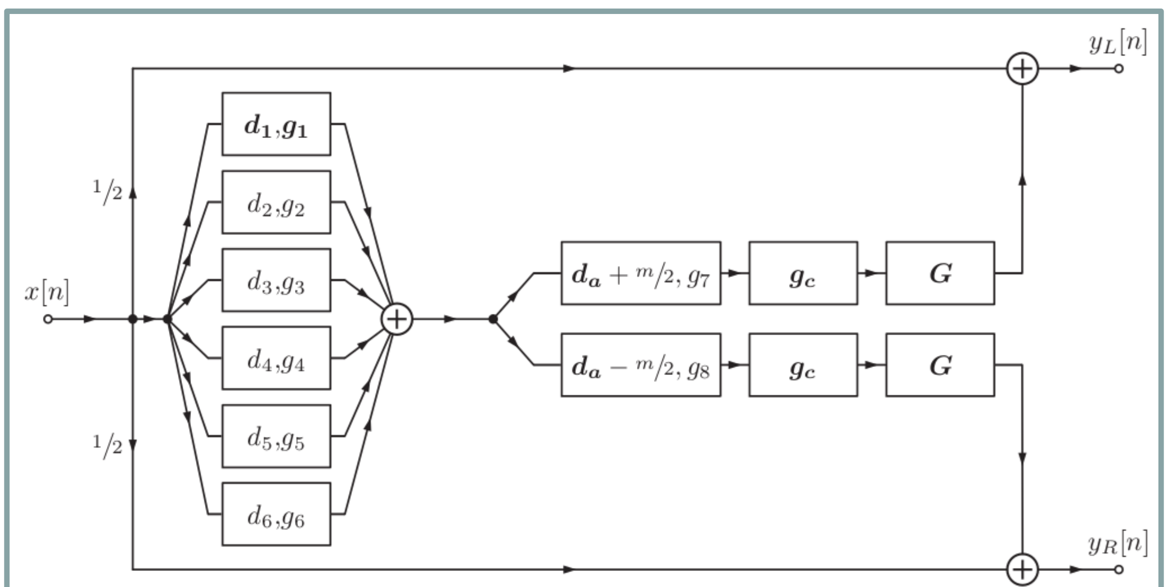
- Estimation of low level filter coefficients from impulse response (IR) characteristics
- Automatic Application of Reverberation on audio files based on previous examples



CIS centre for intelligent sensing

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Automatic artificial reverberation

- IR Parameters (RT60, ED, C, Tc, SC) to filter coefficients (gains & delays)
  - Numerical optimization problem
- Learning by examples
  - Asked 3 experts to apply reverb
  - Supervised Learning (SVMs/HMMs)
    - Input: 31 Audio related features
      - MFCCs, ZCR, CF, …
    - Output: Filter parameters
  - Evaluation: Listening Tests
    - 16 Test Subjects
    - 33 Audio Segments
    - 8 stimuli (inc. reference & anchor)

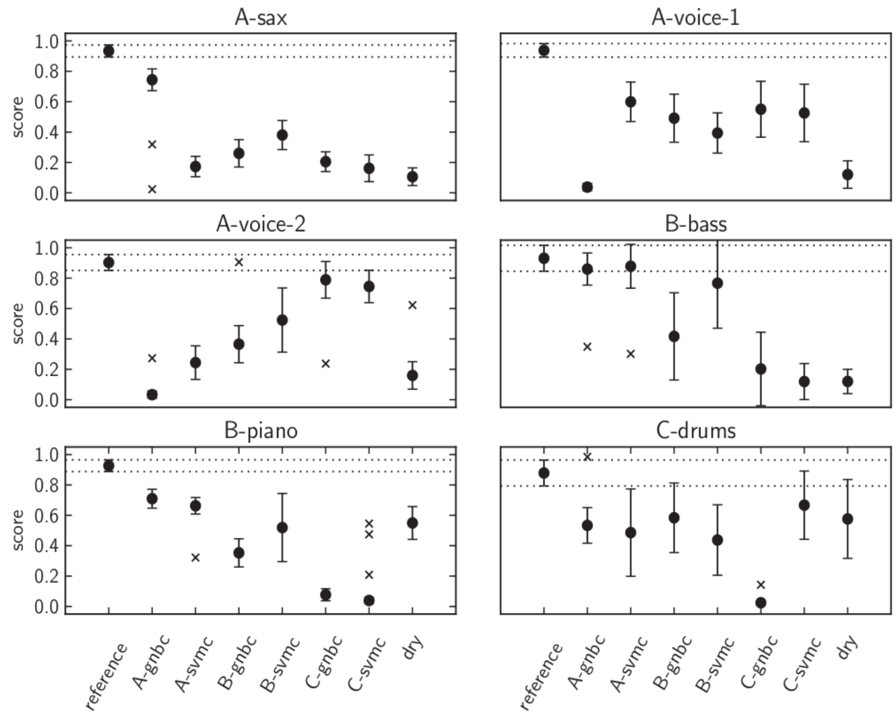
Results and conclusion

Table 6. Weighted f1-scores for the user-trained models. Highest scores for each user are in bold. $|\mathcal{C}|$ is the number of classes calculated for each user.

User	$\|\mathcal{C}\|$	GNB	SVM	HMM	HMM^{SVM}
A	30	**0.79**	0.73	0.06	0.11
B	22	**0.74**	0.66	0.17	0.16
C	32	0.81	**0.84**	0.12	0.18

Table 7. MSEs for the user-trained models. Lowest MSEs for each user are in bold. $|C|$ is the number of classes calculated for each user.

User	$\|\mathcal{C}\|$	GNB	SVM	HMM	HMM^{SVM}
A	30	**0.0104**	0.0138	0.0510	0.0568
B	22	**0.0141**	0.0226	0.0538	0.0386
C	32	**0.0087**	0.0091	0.0444	0.0480



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E. T. Chourdakis and J. D. Reiss,
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