Self-Interference-Cancellation for Full-Duplex Underwater Acoustic Systems

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Outline

- Introduction
- General structure of the proposed digital canceller
- New evaluation metric for the self-interference cancellation (SIC) performance
- Adaptive filtering algorithms for SIC in fast-varying SI channels
- Related work
- Conclusions

Introduction

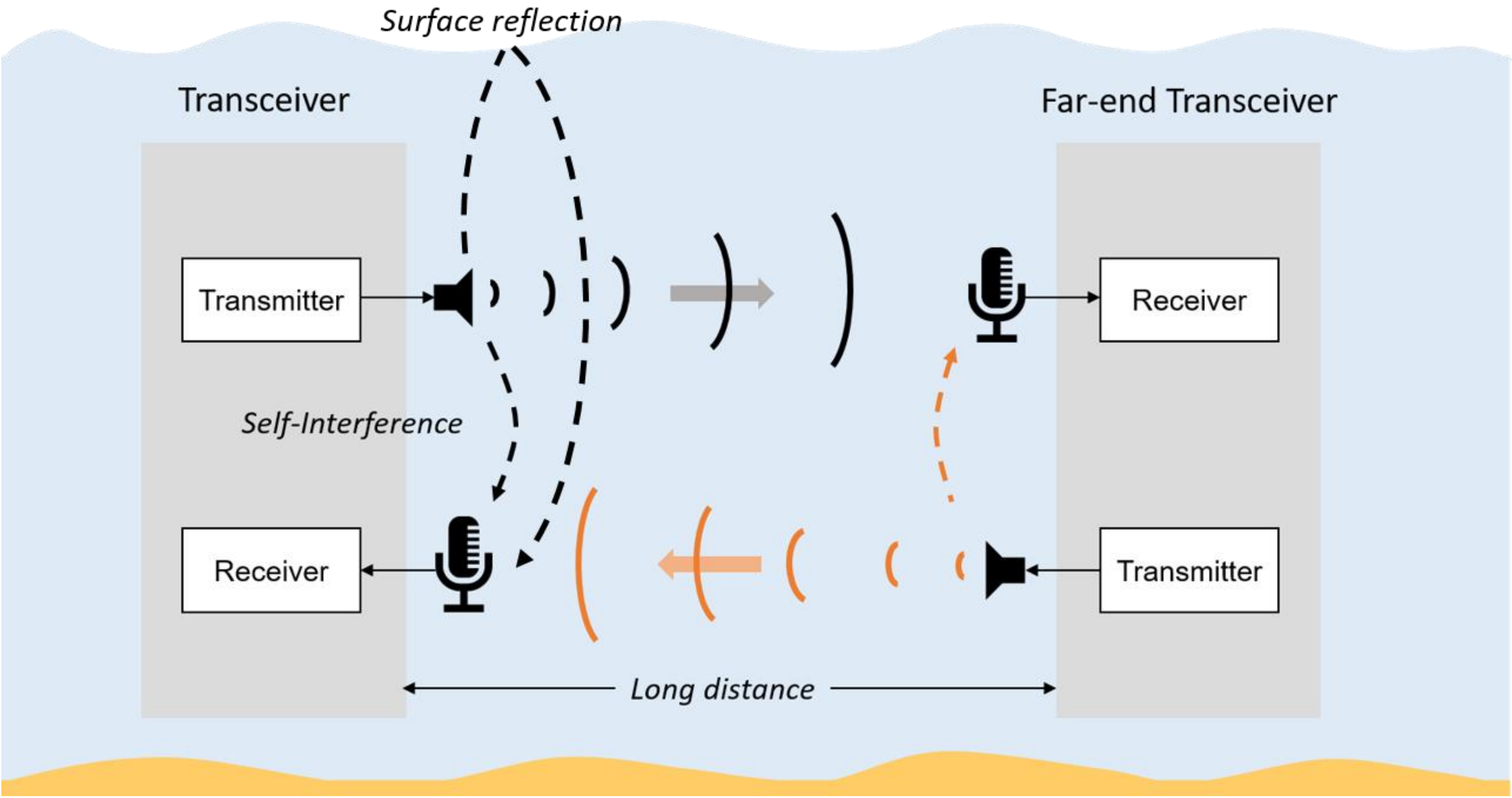
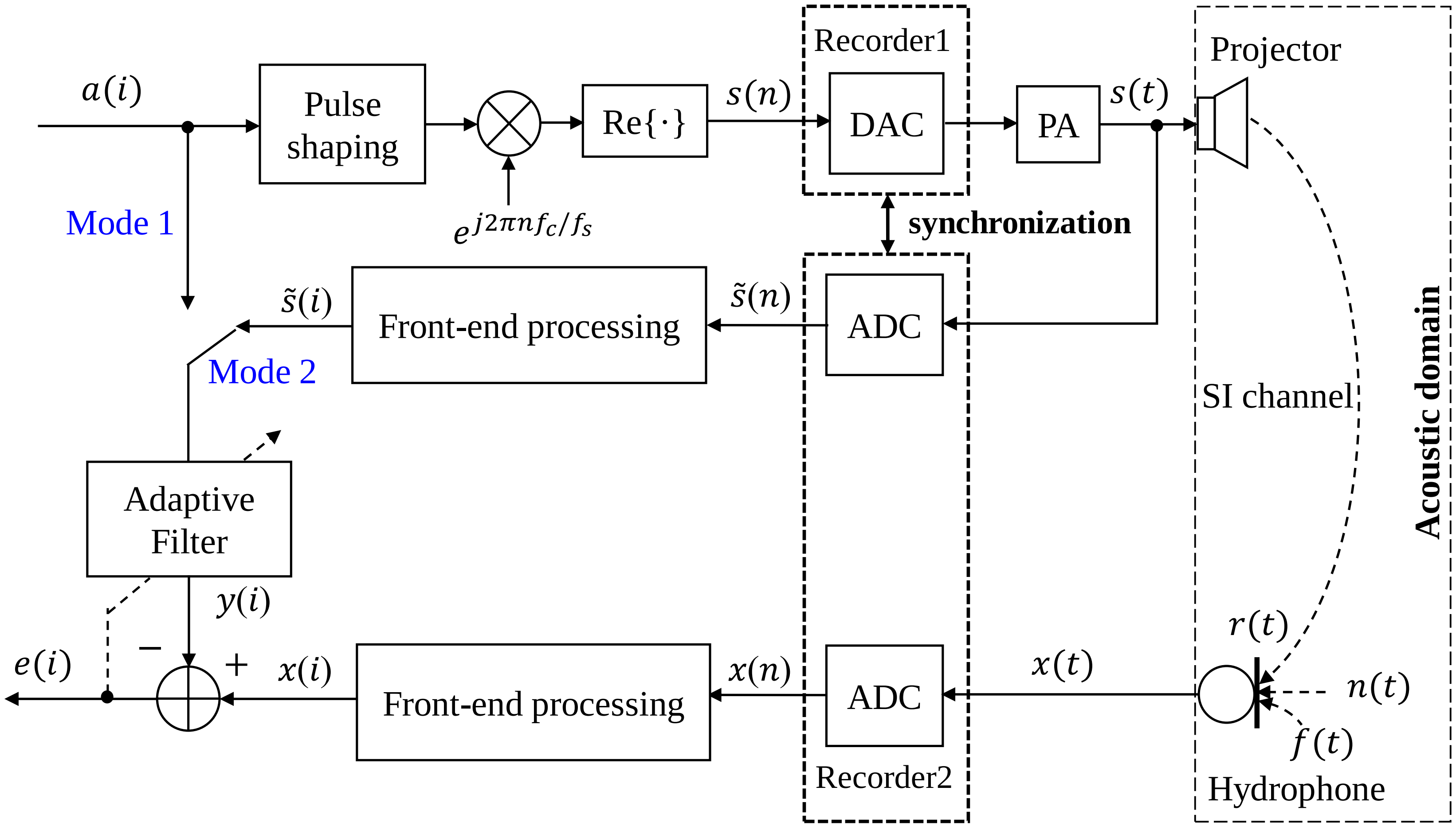


Fig. Illustration of a full-duplex underwater acoustic system

General structure of digital canceller

Digital SI canceller with different reference signals



- Two reference signals are considered for digital SI cancellation (Mode 1: digital data; Mode 2: PA output).

More details in: L. Shen, B. Henson, Y. Zakharov and P. Mitchell, “Digital self-interference cancellation for full-duplex UWA systems," *IEEE Trans on Circuits and Systems II: Express Briefs*, vol. 67, no. 1, pp. 192-196, 2019.

Tank experimental results

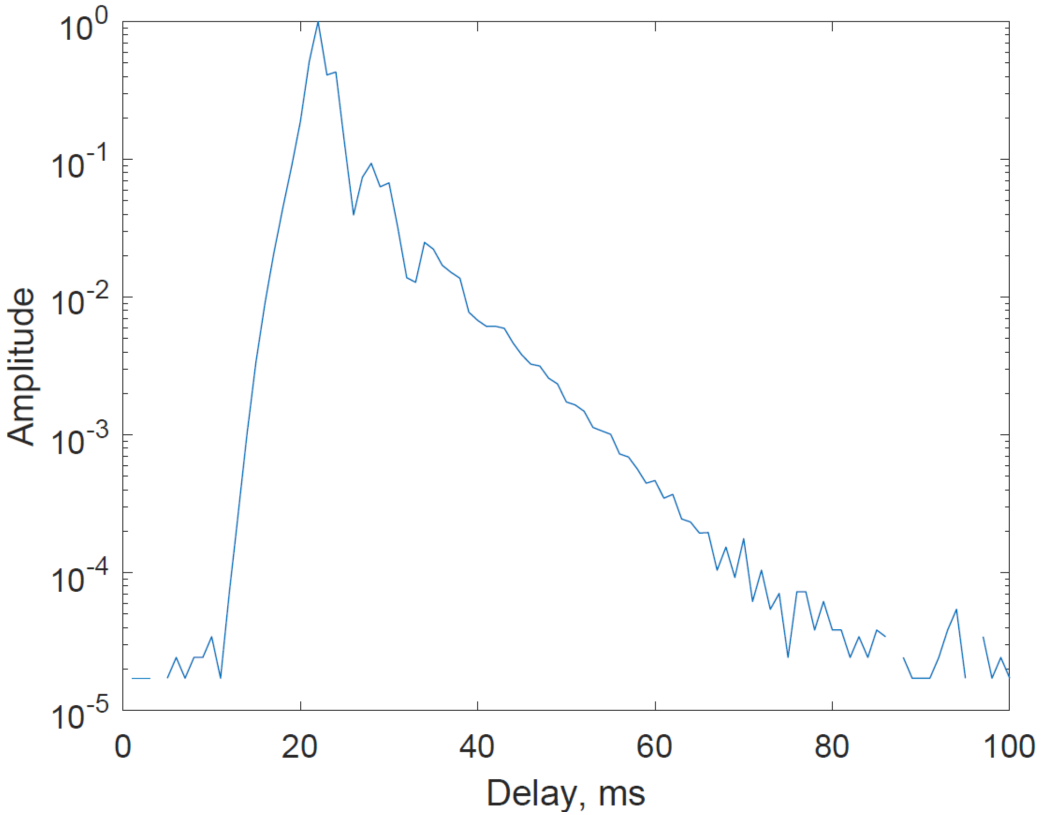


Fig: Impulse response estimate of the SI channel

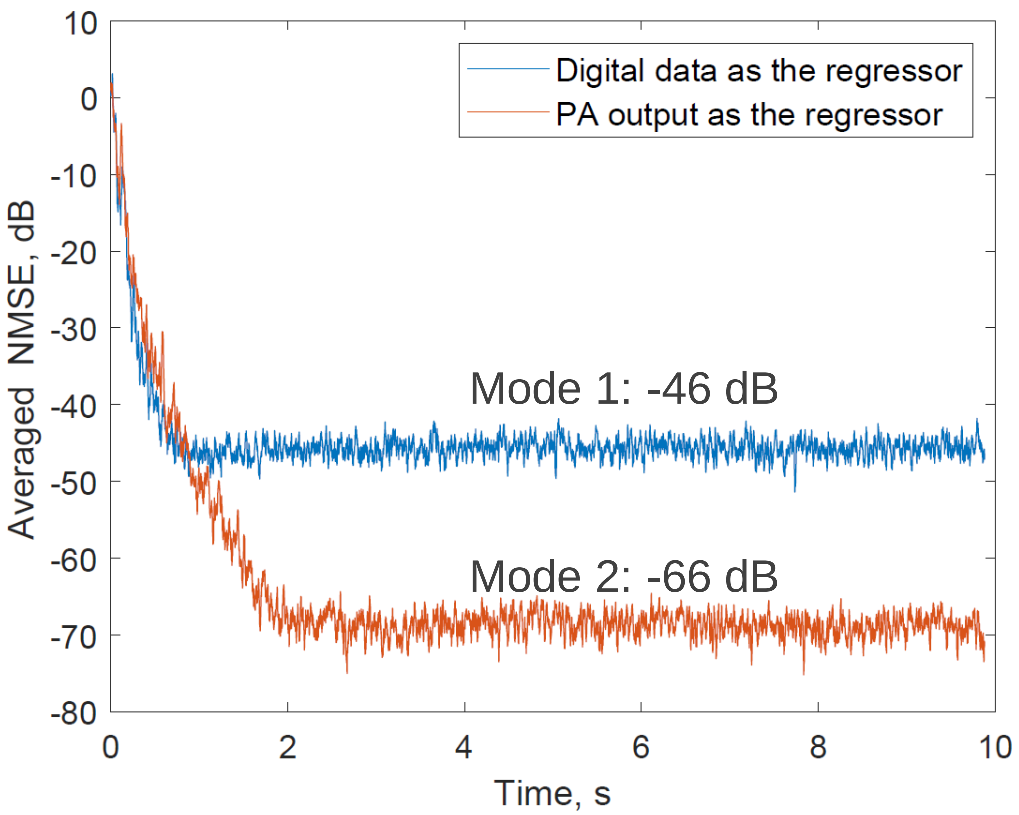
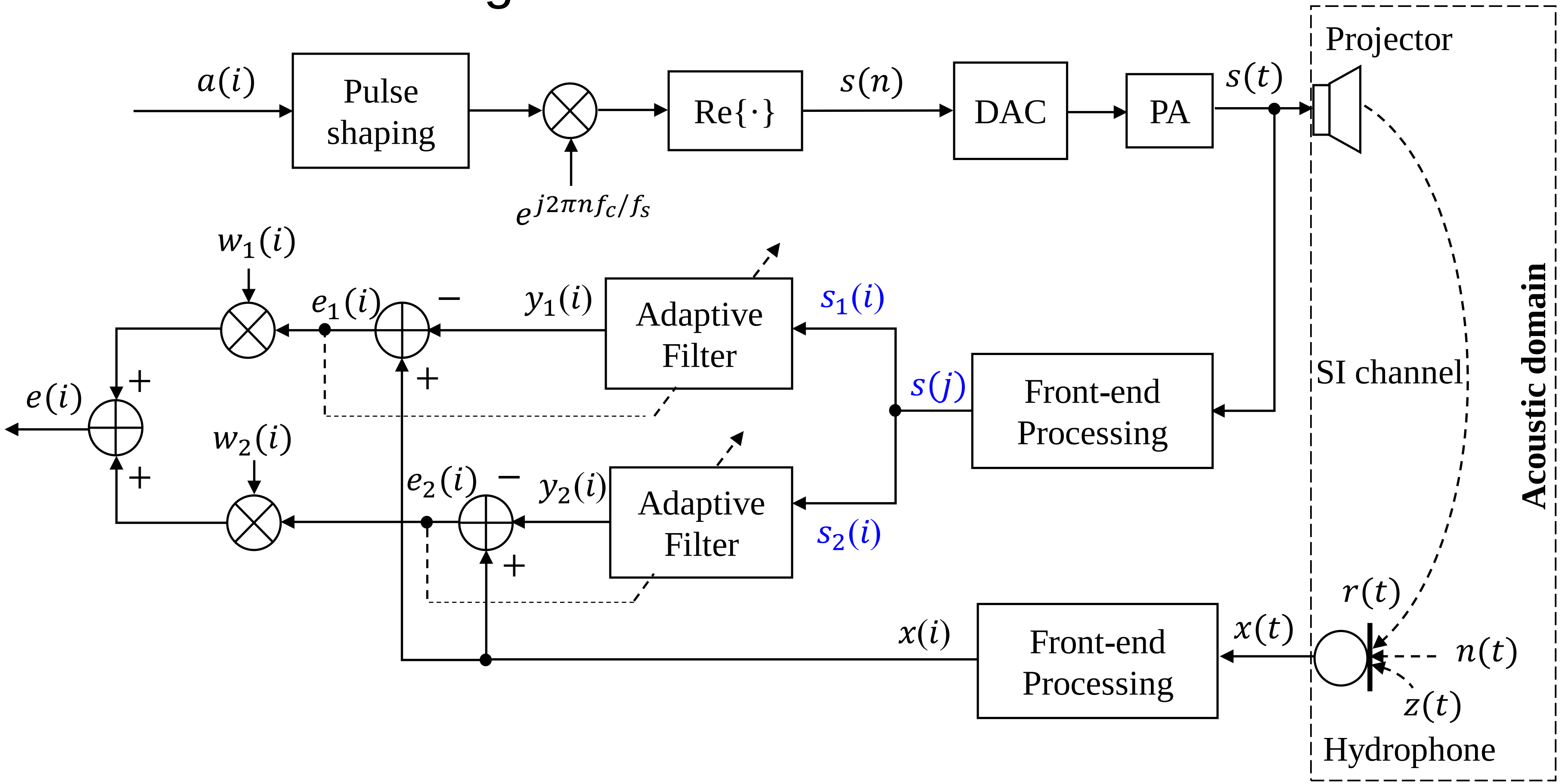


Fig: Averaged NMSE performance of the two SIC modes

Extended structure of the digital SI canceller



- The PA output is down-sampled to twice the symbol rate and interleaved into two branches.
- A small weight coefficient is applied to the branch with high variance (residual signal power), and a large weight coefficient is applied to the branch with high level of SIC.

More details in: L. Shen, B. Henson, Y. Zakharov and P. Mitchell, "Robust digital self-interference cancellation for UWA systems: Lake experiments," in *Underwater Acoustics Conference and Exhibition, Greece*, 2019, pp. 243-250.

Lake experiment

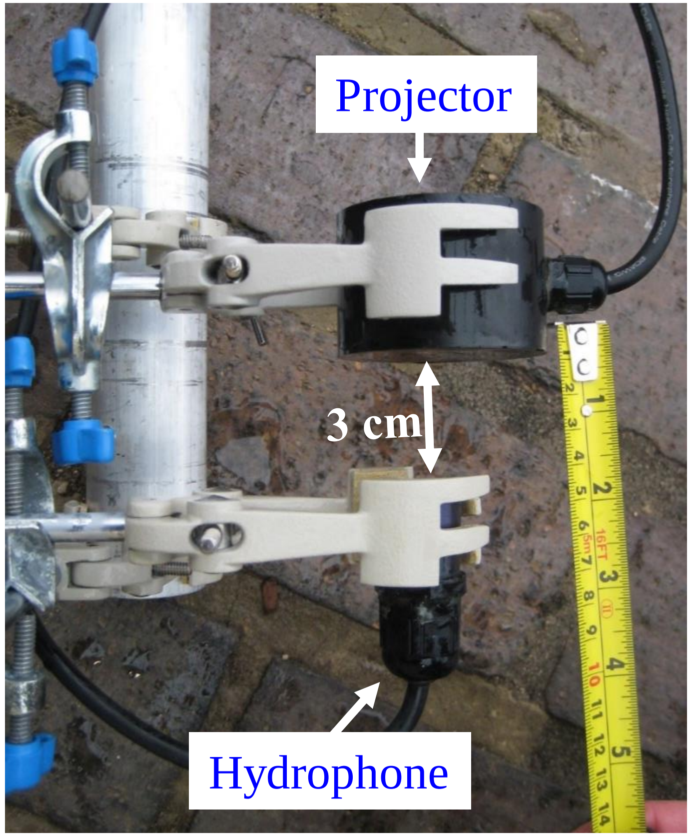


Fig: Experimental setup

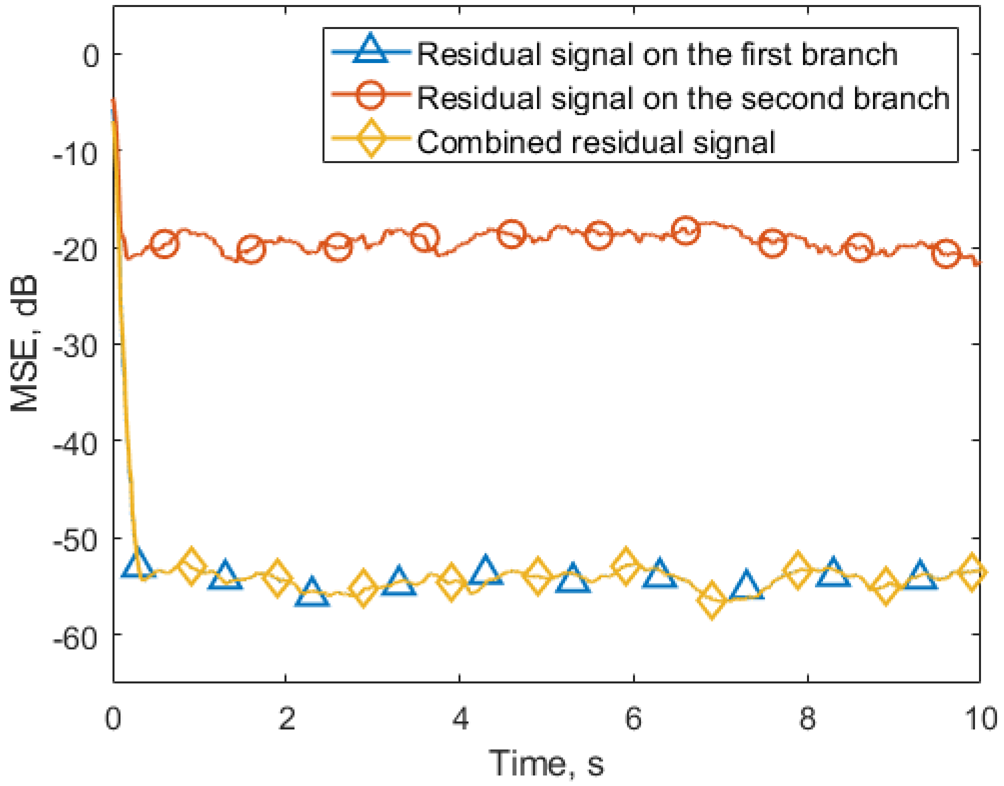
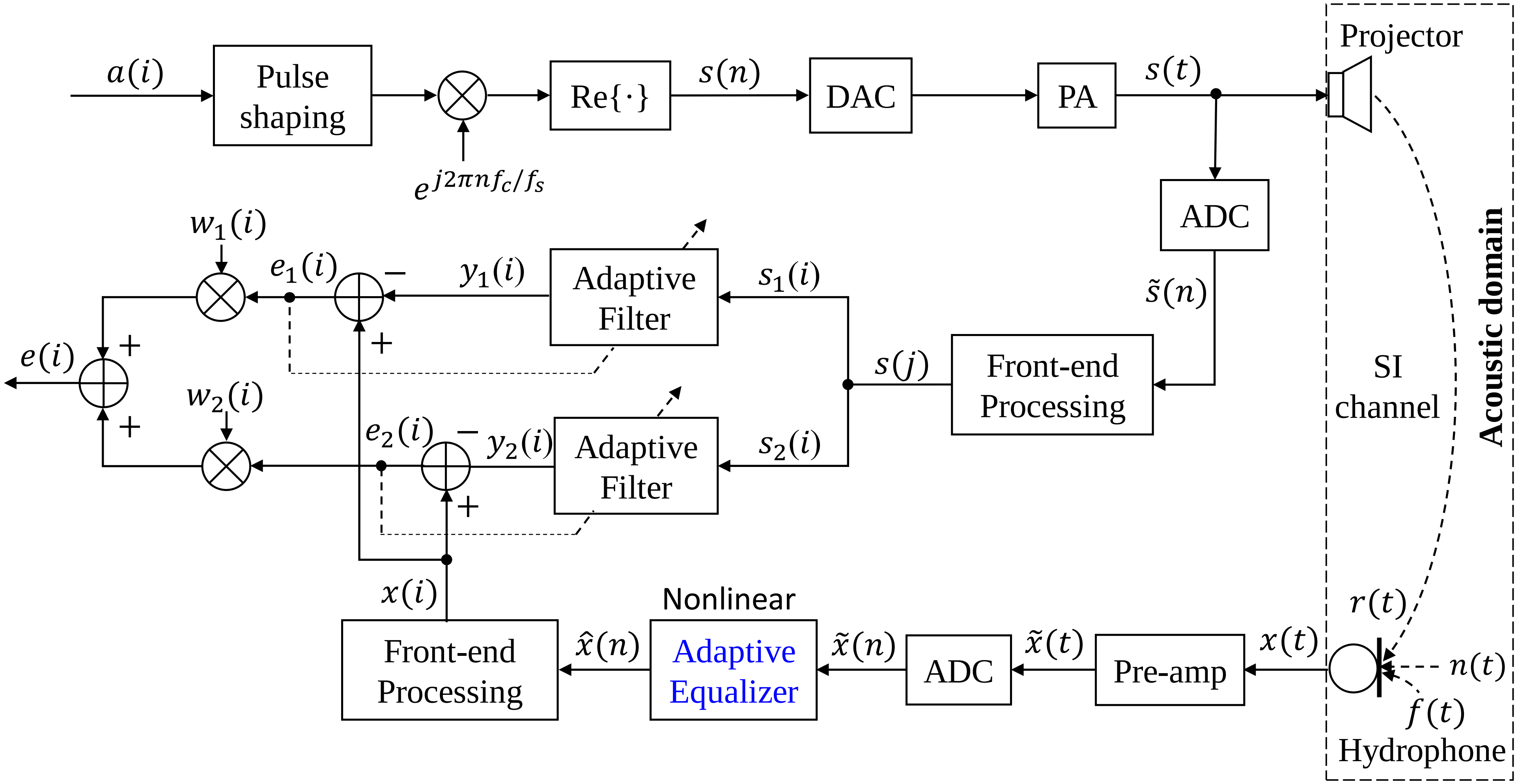


Fig: MSE performance of the lake experiments

Adaptive equalization of the nonlinearity in the hydrophone pre-amplifier



More details in: L. Shen, B. Henson, Y. Zakharov, and P. D. Mitchell, “Adaptive nonlinear equalizer for full-duplex underwater acoustic systems," *IEEE Access*, vol. 8, pp. 108169 – 108178, 2020.

Adaptive filtering algorithms for SIC in fast-varying SI channels

Time-varying self-interference channel

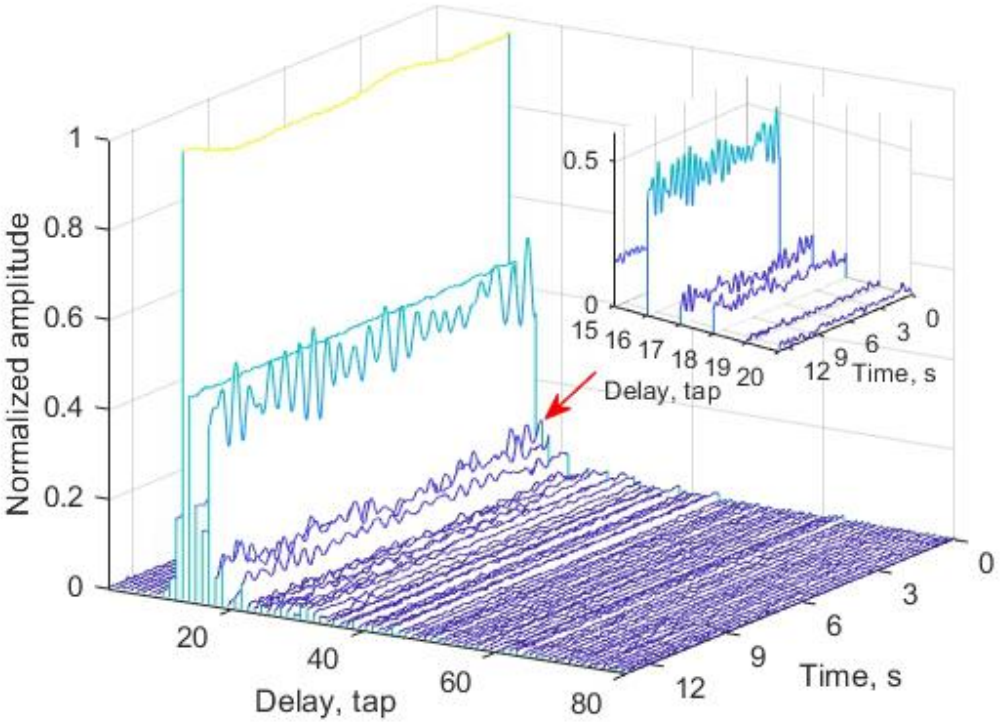


Fig: Time-varying SI channel impulse response in a lake experiment

- Lake depth: 8 m
- The transducer (Tx) and hydrophone (Rx) are placed in the middle of the lake (around 4 m depth).
- The distance between Tx and Rx is 1.3 m.

Evaluation metric

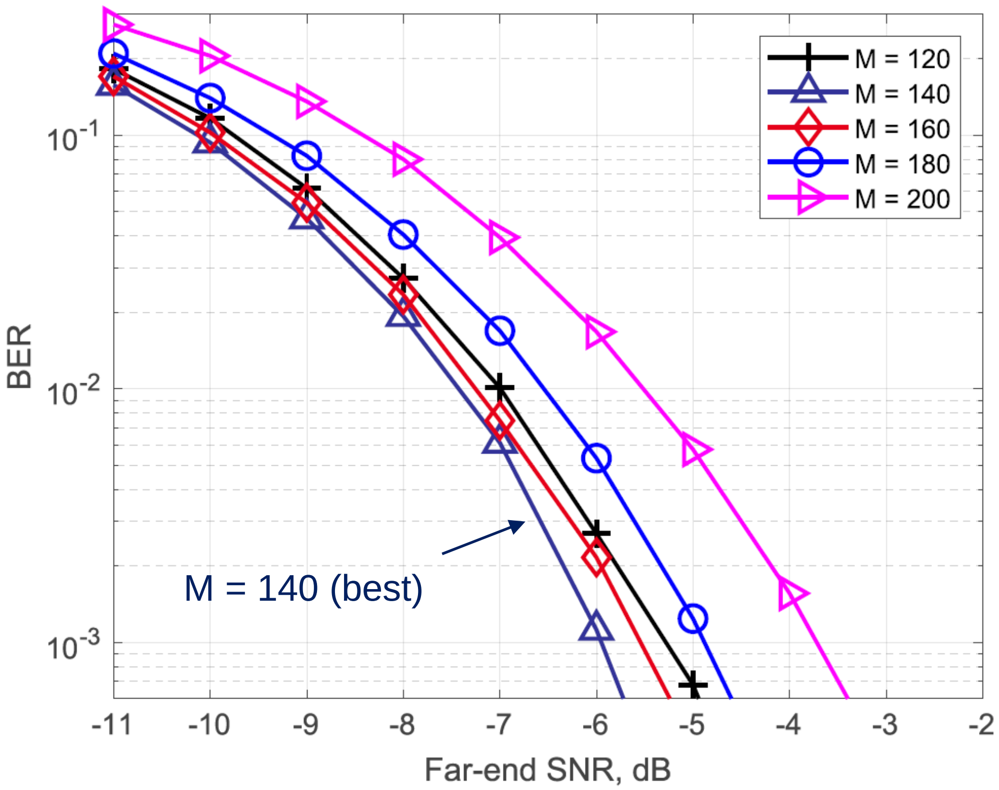
Evaluation of SIC performance

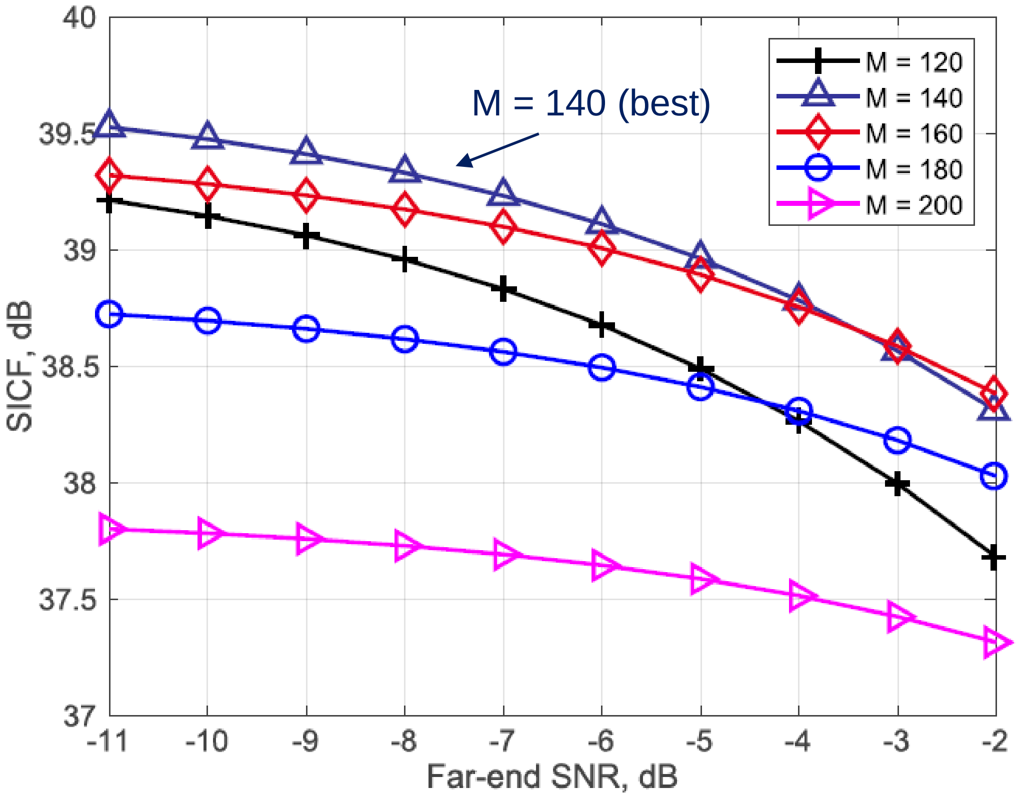
- For Simulation:
  - The mean squared deviation (MSD) is used for evaluating the SIC cancellation performance.
- For lake experiments:
  - For classical adaptive filters, the SIC performance can be evaluated by computing the mean squared error (MSE).
  - For interpolating (non-causal) adaptive filters with improved tracking performance, the MSE cannot accurately represent the SIC performance due to the *over-fitting*.
- The SIC factor (SICF) is proposed, which measures the SIC performance as a factor of improvement in the far-end-signal-to-interference ratio due to the SIC.
- The SICF can be used in practice for both causal and non-causal adaptive filters. It can be used to adjust the parameters of the adaptive algorithms for SIC without implementing a whole FD system.

More details in: L. Shen, Y. Zakharov, B. Henson, N. Morozs and P. Mitchell, “Adaptive filtering for full-duplex UWA systems with time-varying self-interference channel," *IEEE Access*, vol. 8, pp.187590-187604, 2020.

SICF and BER

- We compare the SICF and the BER performance provided by the SI canceller in fast-varying channel using the SRLS-P adaptive filter with different sliding window length M.





More details in: L. Shen, Y. Zakharov, B. Henson, N. Morozs and P. Mitchell, “Adaptive filtering for full-duplex UWA systems with time-varying self-interference channel," *IEEE Access*, vol. 8, pp.187590-187604, 2020.

SRLSd and SRLS-P adaptive filters

- SRLS adaptive filter with a delay (SRLSd)
  - A delay is introduced between $\hat{h}(i)$ and $s(i)$ to improve the tracking performance.
  - In fast-varying channels, the tracking performance is still limited.
- SRLS-P adaptive filter
  - Exploit parabolic approximation of the time-varying channel response.
  - It improves the tracking performance at the expense of high complexity.

More details in: L. Shen, Y. Zakharov, B. Henson, N. Morozs and P. Mitchell, "Adaptive filtering for full-duplex UWA systems with time-varying self-interference channel," *IEEE Access*, vol. 8, pp.187590-187604, 2020.

Adaptive filtering based on basis expansion model

- Low-complexity interpolating adaptive filters are proposed based on BEM and weighted least square (LS) approach.
- As an example, we use the Legendre polynomials as basis functions and propose the SRLS-L adaptive filter.
- Advantages:
  - No limitation on the choice of basis functions
  - Suitable for complex-valued data
  - Low-complexity (iterative computation + FFTs + DCD algorithm)
- Limitation:
  - For fast-varying channels with a large delay spread, the minimum sliding window length required is significantly increased when high orders of the basis functions are used ($M > (P+1)L$).

More details in: L. Shen, Y. Zakharov, L. Shi, and B. Henson, “Adaptive Filtering Based on Legendre Polynomials”, *TechRxiv. Preprint*. https://doi.org/10.36227/techrxiv.13084460.v1

HSRLS-L-DCD adaptive filter

- To exploit the sparsity in the expansion coefficients, the homotopy SRLS-L-DCD (HSRLS-L-DCD) adaptive filter is proposed.
- In the HSRLS-L-DCD adaptive filter, a solution is found by minimizing a cost function which contain the LS cost and a penalty function that attracts sparsity.

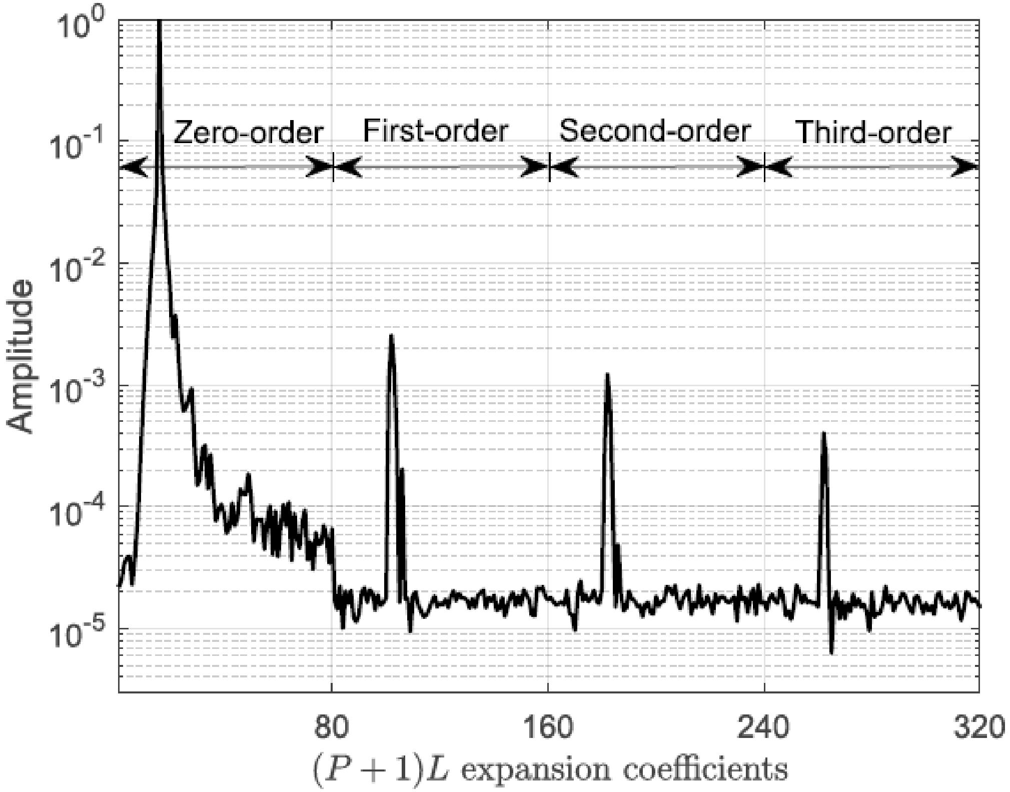


Fig: Estimates of expansion coefficients

More details in: L. Shen, Y. Zakharov, L. Shi, and B. Henson, " BEM Adaptive Filtering for SI Cancellation in Full-Duplex Underwater Acoustic Systems.", *submitted to Signal Processing*, Under review.

Simulation scenario

The SI channel is simulated based on the SI channel estimate from a lake experiment.

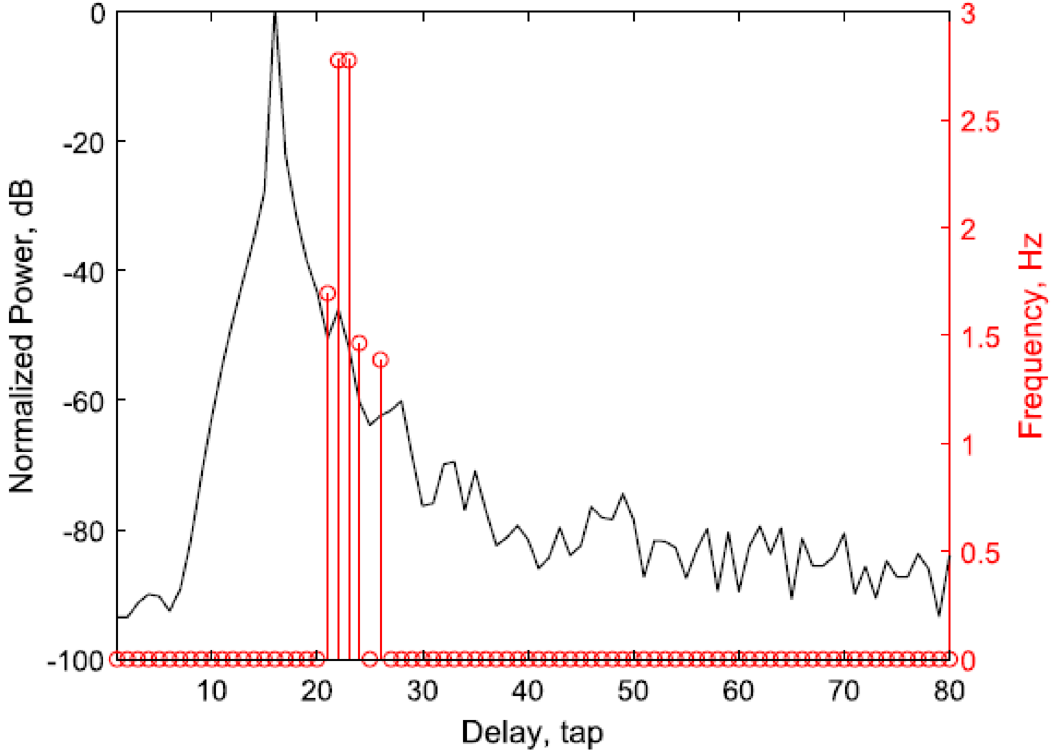


Fig: Power delay profile and cut-off frequency of the multipath components in the FD experiment

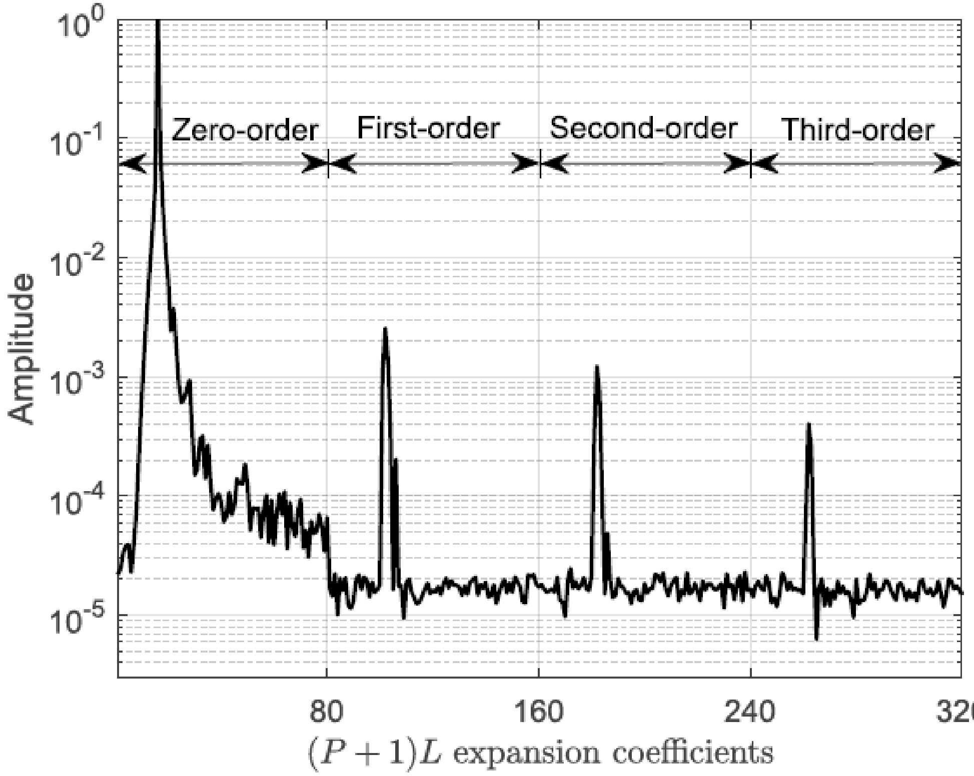


Fig: Estimates of expansion coefficients

Simulation results

- MSD is used to evaluate the channel identification performance.
- The SI to noise ratio is 71 dB.

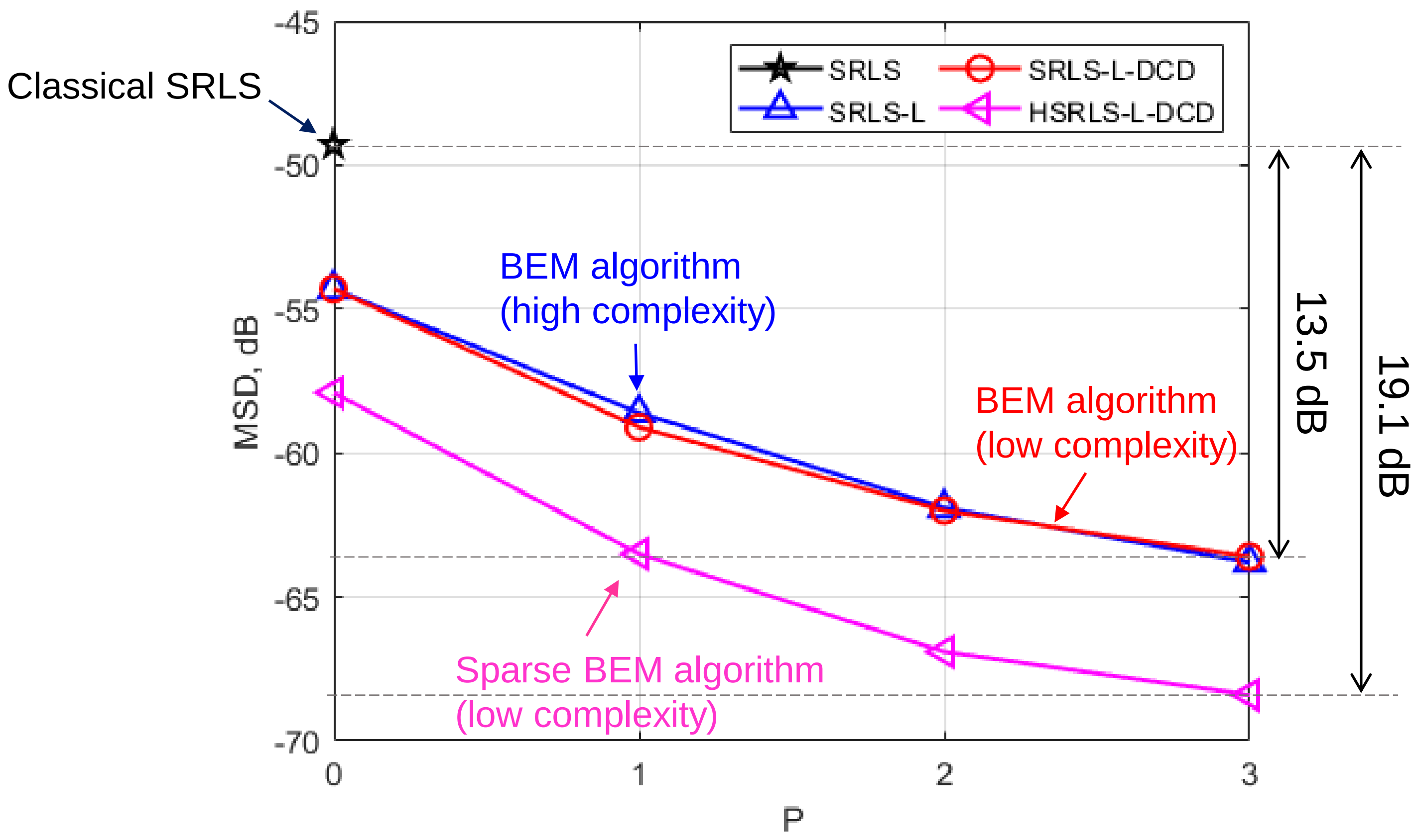
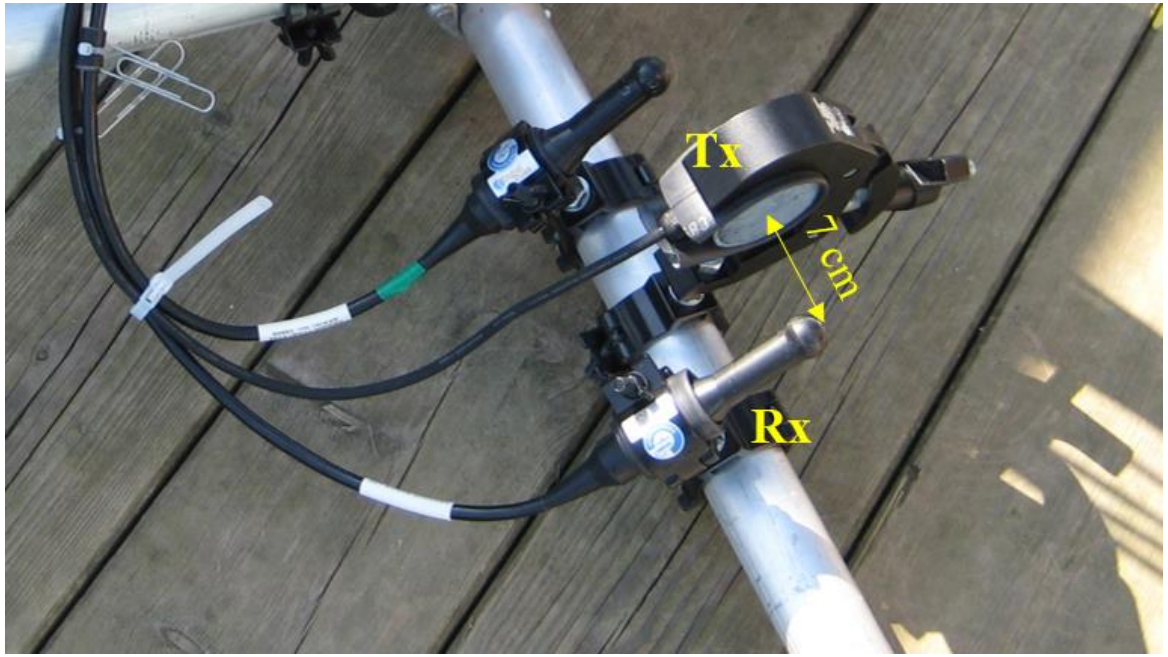
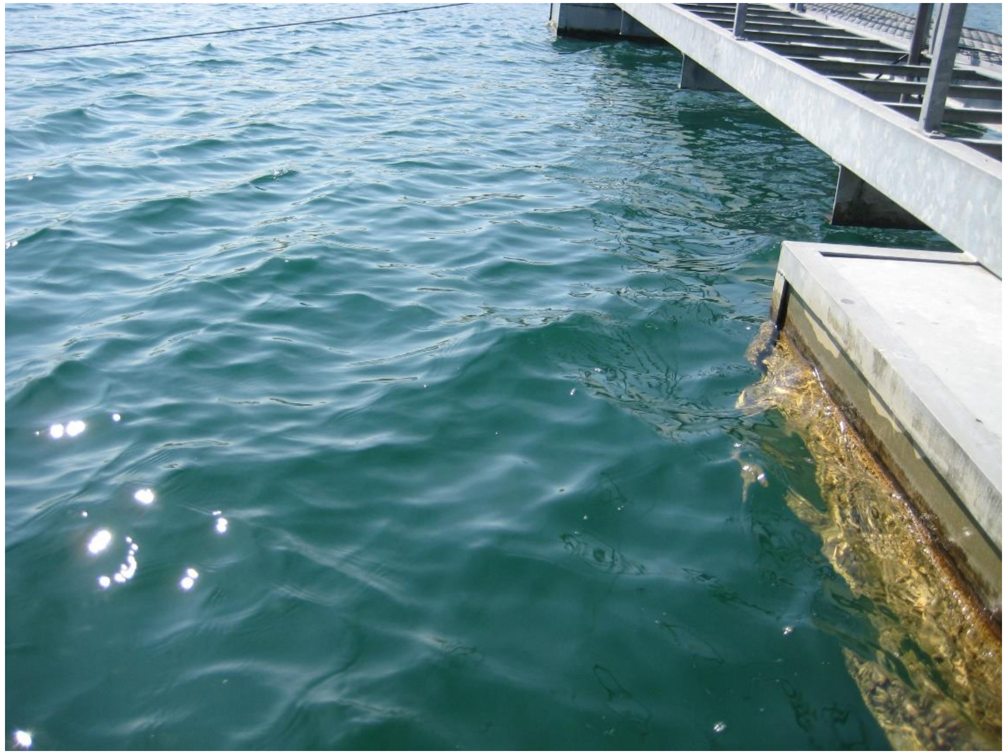


Fig: MSD performance of the adaptive algorithms

Lake experiment





- The Tx and Rx are positioned at a depth of 4 m.
- The lake depth is around 8 m.
- During the experiment, the amplitude of the lake surface waves varies from 5 cm to 10 cm.

Experimental results

Carrier frequency: 32 kHz   Filter length: 80 taps

Adaptive filter	Sliding window length M	SIC, dB	Improvement compared to SRLS, dB
SRLS	105	51.2	0
SRLS-L, P = 0	185	55.5	4.3
SRLS-L, P = 1	225	57.7	6.5
SRLS-L, P = 2	325	58.9	7.7
SRLS-L, P = 3	385	59.7	8.5
HSRLS-L-DCD, P = 0	105	57.3	6.1
HSRLS-L-DCD, P = 1	105	60.9	9.7
HSRLS-L-DCD, P = 2	165	62.3	11.1
HSRLS-L-DCD, P = 3	225	63.4	12.2

Related work

- Acoustic-domain SIC scheme with two projectors

  The SIC is performed using an extra (secondary) projector that emits an acoustic signal for cancelling the SI at the receive antenna.

- Two-stage digital SIC scheme with two hydrophones

  First stage: To cancel the strong and stable SI signal from the direct path; Second stage: Adaptive beamforming is used to cancel the time-varying reflections from the sea surface.

- Investigate the bit error rate performance of the whole FD system with both near-end and far-end transmission (on-going work)
  - Apply the proposed interpolating adaptive filters with good tracking ability.
  - Further improve the SIC performance by jointly estimate the near-end and far-end channels in turbo iterations.

More details in: Y. Wang, Y. Li, L. Shen, and Y. Zakharov, “Acoustic-domain self-interference cancellation for full-duplex underwater acoustic communication systems," in *IEEE Asia-Pacific Signal and Information Processing Association Annual Summit and Conference*, 2019, pp. 1112-1116.
L. Shen, B. Henson, Y. Zakharov and P. Mitchell, “Two stage self-interference cancellation for full-duplex UWA systems,“ in *MTS/IEEE Oceans, Marseille, France*, June 2019.

Conclusions

- We propose various SIC techniques for FD UWA systems, most of the which are digital cancellation based on adaptive filtering.
- Regarding the fast channel variation due to the moving lake/sea surfaces, two approaches have been proposed; one is to use interpolating adaptive filtering algorithms which are capable of tracking the fast-varying channels, the other approach is to use multiple hydrophones for adaptive beamforming.
- An acoustic-domain SIC scheme using multiple projectors has been proposed to achieve extra amount of SIC in the acoustic domain.
- Based on the experimental results, it can be concluded that a high level of SIC has been achieved with the proposed SI canceller structure and novel adaptive filtering algorithms.