

CHANNEL ESTIMATION FOR CROSSTALK CANCELLATION IN WIRELESS ACOUSTIC NETWORKS



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1. Crosstalk Canceller (CC) over a WASN

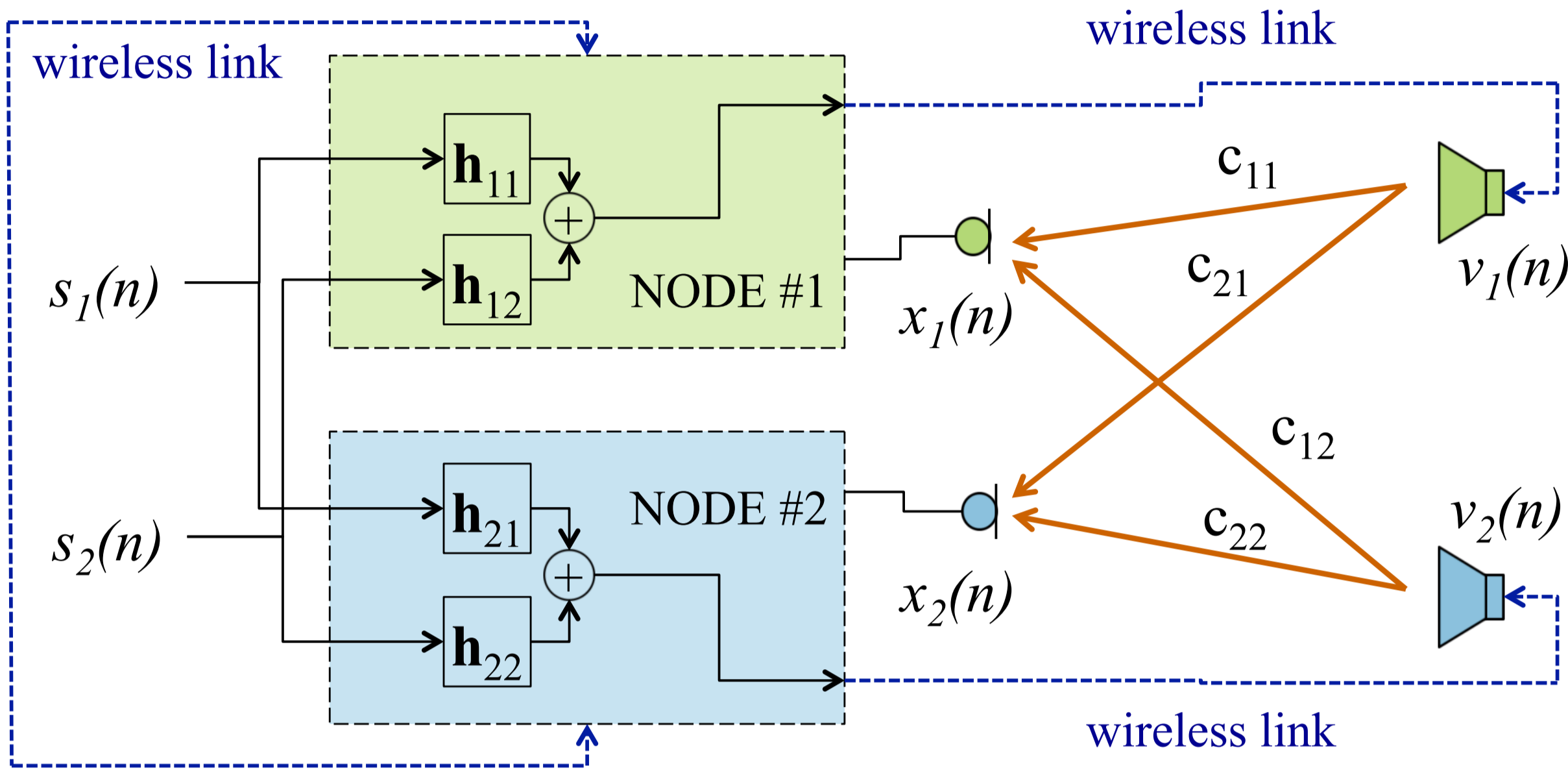
WASN: Wireless Acoustic Sensor Network

ORIGINAL SOUNDS

- $s_1(n)$: (e.g. Male speech)
- $s_2(n)$: (e.g. Female speech)

CC DESIRED RESPONSE

- $x_1(n) \approx s_1(n)$
- $x_2(n) \approx s_2(n)$



Step 1) > Estimate the **electro-acoustic channels** (also called Room Impulse Responses, **RIR**) c_{ij} using maximum length sequences (MLS) [1].

$$c_{ij} = [c_{ij}(0) \quad c_{ij}(1) \quad \dots \quad c_{ij}(L_c - 1)]^T$$

Step 2) > Design the **CC filters** h_{ij} [2].

Step 3) > Filter original signals $s_1(n)$ and $s_2(n)$ through filters h_{ij} to obtain $v_1(n)$ and $v_2(n)$. Provide the desired signals $x_1(n)$ and $x_2(n)$ at the mic locations.

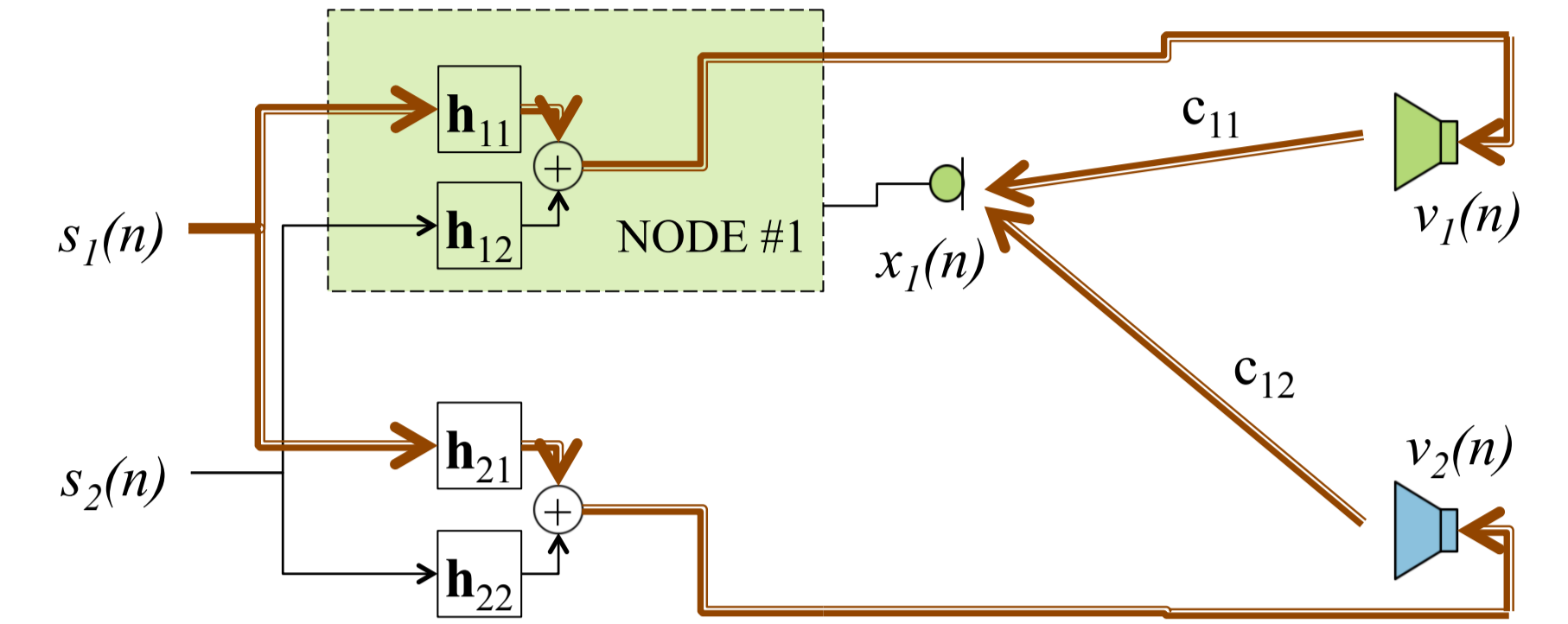
Some challenges due to the use of WASNs:

- Lack of perfect synchronization between the nodes [3].
- Requirement of low communication burden between nodes, thus, no exchange of signals $x_i(n)$ or $v_i(n)$ is allowed.
- **The acoustic channels c_{ij} can vary their coefficients due to changes in the location of the nodes.**

[1] J. Vanderkooy, "Aspects of MLS measuring systems," J. Audio Eng. Soc., vol. 42, no. 4, pp. 219-231, 1994.
 [2] O. Kirkeby et al., "Fast deconvolution of multichannel systems using regularization," IEEE Trans. on Speech and Audio Processing, vol. 6, no. 2, pp. 189-194, Mar 1998.
 [3] G. Piñero et al., "Sound-field reproduction system over a two-node acoustic network of mobile devices," Proc. IEEE 2nd World Forum on Internet of Things (WF-IoT), Milan, 2015, pp. 652-657.

2. Adaptive identification of the acoustic channels

Let us define the **Global Impulse Response (GIR)** a_{11} of **NODE #1** as the impulse response between $s_1(n)$ and $x_1(n)$:



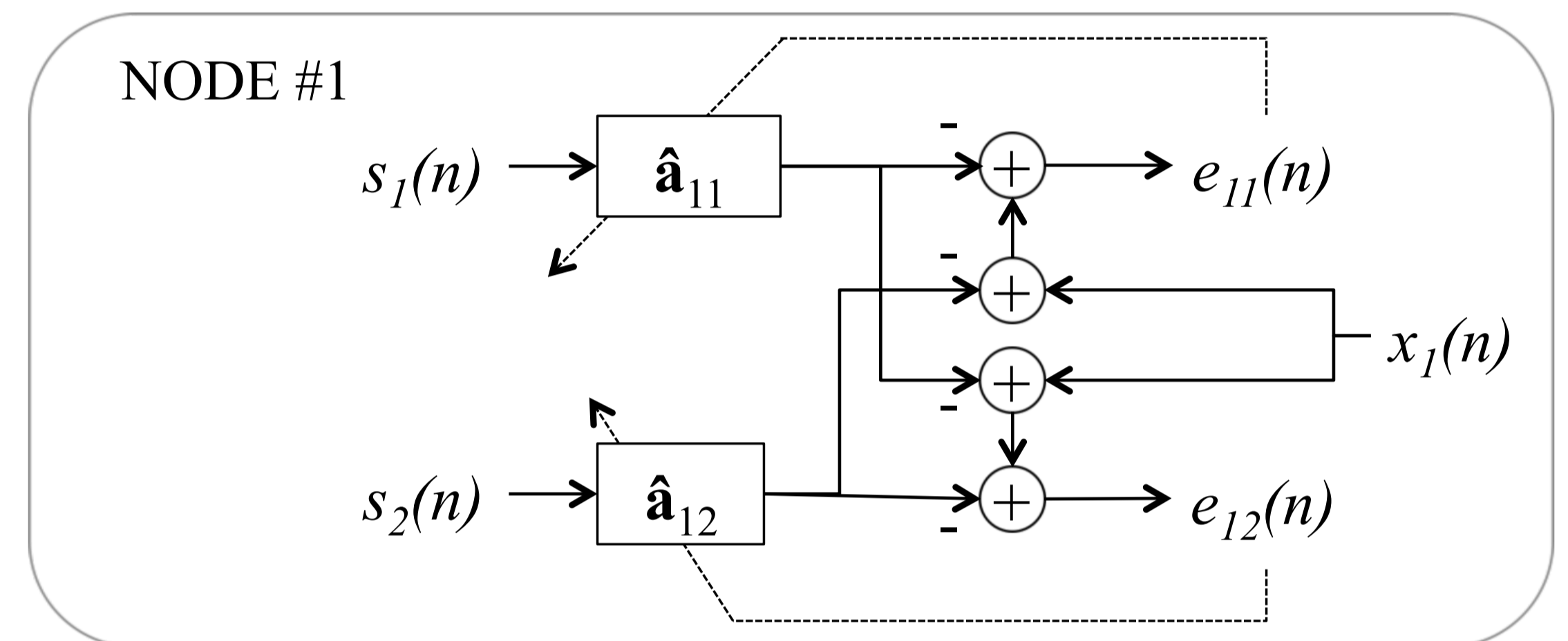
$$a_{11} = c_{11} * h_{11} + c_{12} * h_{21}$$

We define in the same way the GIR between $s_2(n)$ and $x_1(n)$ as:

$$a_{12} = c_{11} * h_{12} + c_{12} * h_{22}$$

Therefore: $x_1(n) = a_{11} * s_1(n) + a_{12} * s_2(n)$

Adaptive estimation of the GIRs



Step 1) > The estimation of the GIRs related to **NODE #1** is carried out minimizing the mean square of the following error signals:

$$e_{11}(n) = [x_1(n) - \hat{a}_{12} * s_2(n)] - \hat{a}_{11} * s_1(n)$$

$$e_{12}(n) = [x_1(n) - \hat{a}_{11} * s_1(n)] - \hat{a}_{12} * s_2(n)$$

Similar procedure to estimate the GIRs associated to **NODE #2** where:

$$a_{21} = c_{21} * h_{11} + c_{22} * h_{21} \quad , \quad a_{22} = c_{21} * h_{12} + c_{22} * h_{22}$$

Step 2) > Once the GIRs have been estimated, the corresponding RIRs are estimated at each node through a least squares (LS) solution.

Step 3) > Follow steps 2) and 3) of the CC algorithm to design the new filters and provide signals $v_1(n)$ and $v_2(n)$ to the loudspeakers.

Implemented algorithms

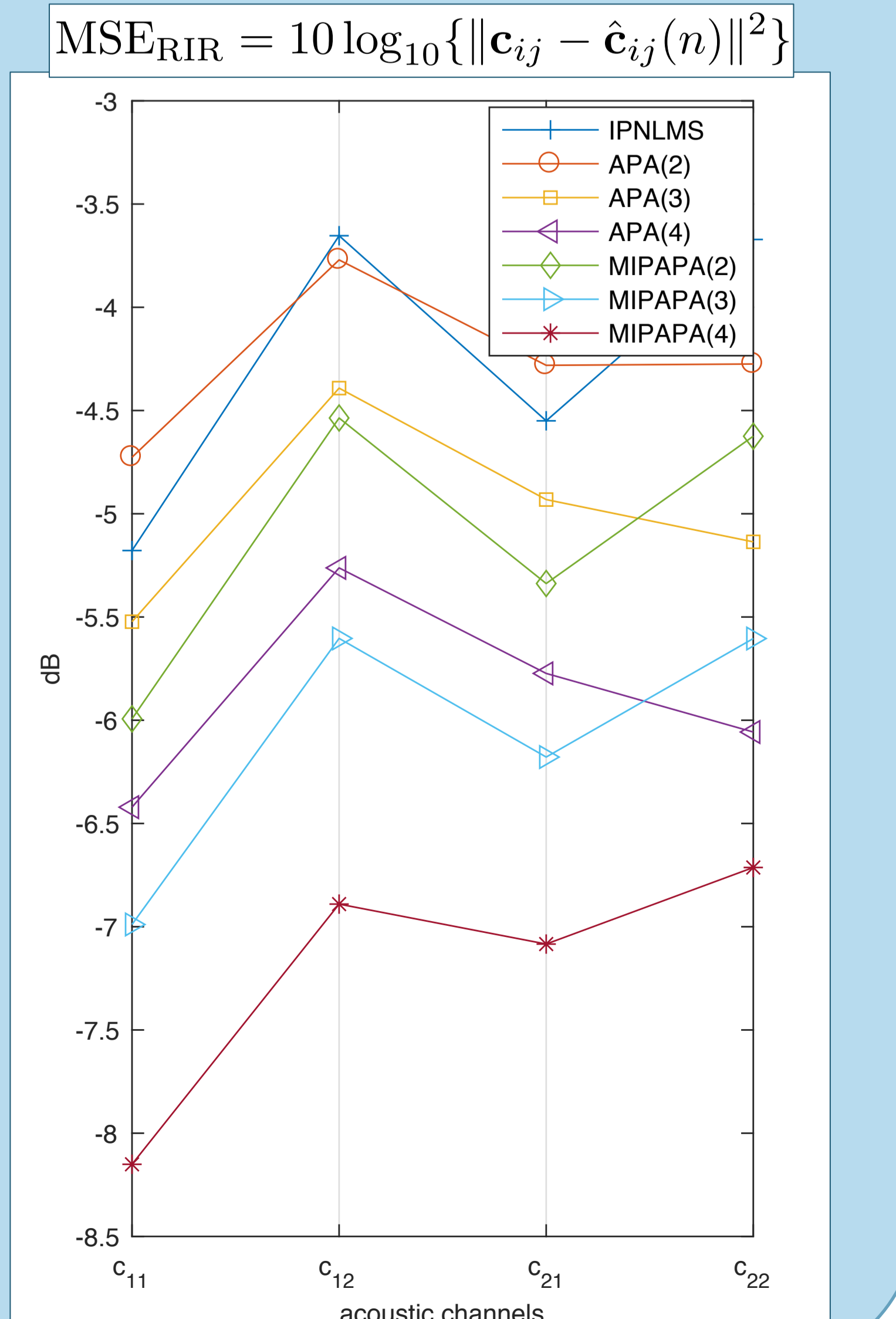
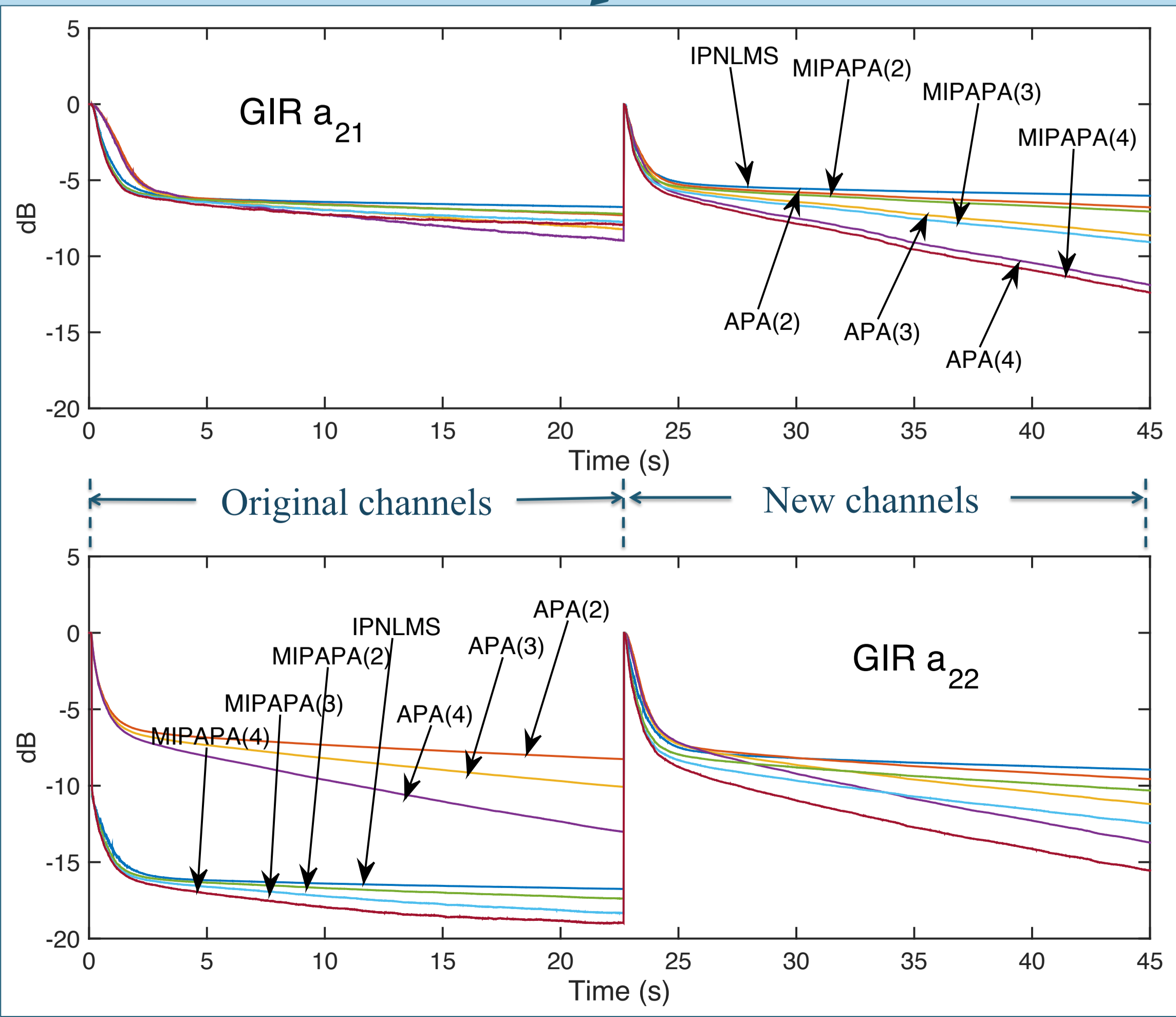
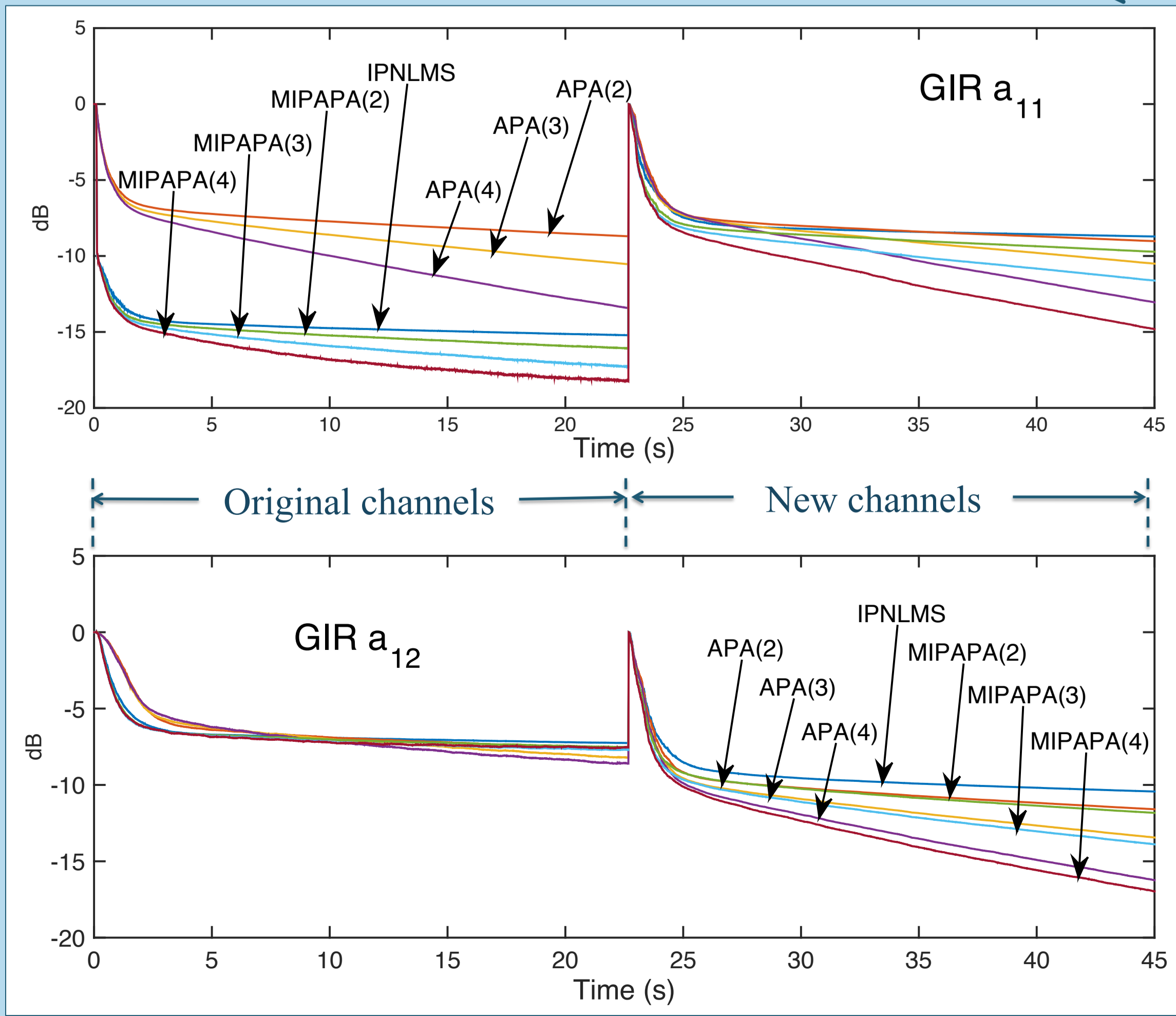
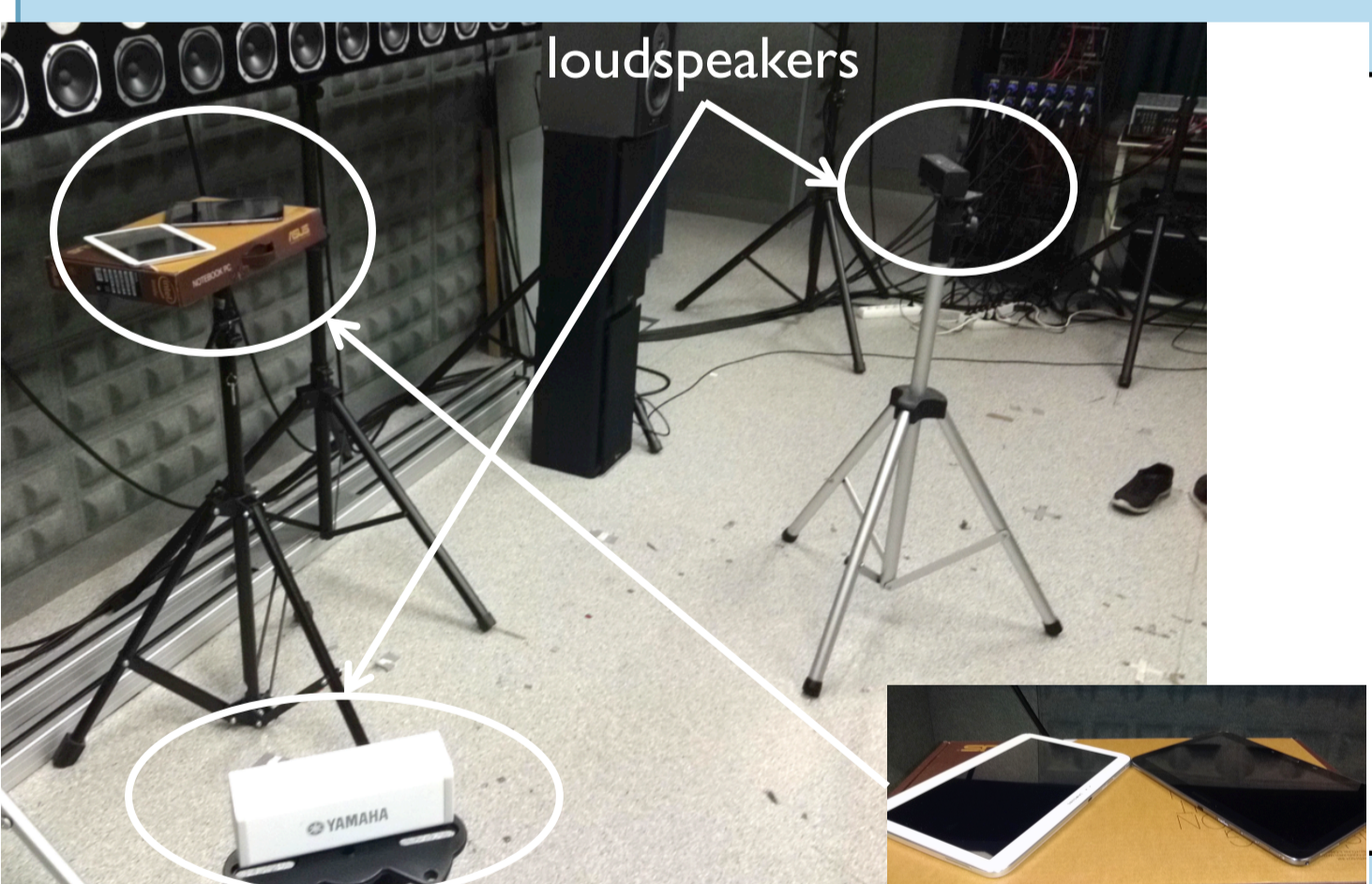
- Affine projection algorithm (APA), $N=2,3,4$
- Improved proportionate NLMS (IPNLMS)
- Memory-improved proportionate APA (MIPAPA), $N=2,3,4$

For sparse GIRs

$$\text{Misalignment} = 20 \log_{10} \frac{\|a_{ij} - \hat{a}_{ij}(n)\|_2}{\|a_{ij}\|_2}$$

3. Simulation Results

- Real acoustic channels measured between two Bluetooth loudspeakers & two tablets (Android OS).
- Number of RIR coefficients: $L_c=1200$.
- Sampling frequency $f_s=11025$ Hz.
- $s_1(n)$ and $s_2(n)$ are uncorrelated white noise.



Conclusions

- GIR Estimation: MIPAPA of order N outperforms the corresponding APA of the same order.
- RIR Estimation: the MSE of MIPAPA of order $N-1$ is similar to that of APA of order N .