MULTIPATH-ENABLED PRIVATE AUDIO WITH NOISE

Anadi Chaman*, Yu-Jeh Liu*, Jonah Casebeer†, Ivan Dokmanić*

Departments of *Electrical and Computer Engineering and †Computer Science
University of Illinois at Urbana-Champaign

ABSTRACT

We address the problem of privately communicating audio messages to multiple listeners in a reverberant room, using a set of loudspeakers. We propose two methods based on emitting noise. In the first method, the loudspeakers emit noise signals that are appropriately filtered so that after echoing along the multiple paths in the room, they sum up and descramble to yield distinct meaningful audio messages only at specific focusing spots, while being incoherent everywhere else. In the second method, adapted from wireless communications, we project noise signals onto the nullspace of the MIMO channel matrix between the loudspeakers and listeners. Loudspeakers reproduce a sum of the projected noise signals and intended messages. Again because of the echoes, the MIMO nullspace changes between the different locations in the room. Thus the listeners at focusing spots hear intended messages, while the acoustic channel of an eavesdropper at any other location is jammed. We show using both numerical and real experiments, that with a small number of speakers and a few impulse response measurements, audio messages can be indeed be communicated to a set of listeners while ensuring negligible intelligibility elsewhere.

Index Terms—Private audio communication, speech privacy, multi-channel convolutional synthesis, speech intelligibility.

1. INTRODUCTION

Consider the problem of sending audio messages to different listeners in a reverberant room, while making sure that each message can only be understood by its intended recipient. Importantly, no eavesdropper anywhere in the room should be able to understand any of the messages.

This problem is related to personal audio zones and sound field reproduction [1, 2, 3, 4, 5, 6, 7, 8, 9] where the goal is to reproduce different sound streams in a few predefined zones in a room, while minimizing the sound level everywhere else. In most of these approaches, however, an eavesdropper with a sensitive microphone (or a good ear) can easily understand the messages. The reason is that the loudspeakers simply reproduce linearly filtered versions of desired messages which remain highly correlated with any residual error signal.

To address the problem of private audio communication, we propose two methods. The first approach communicates audio messages at intended focusing spots by emitting appropriately filtered white Gaussian noise signals from loudspeakers. The filters are constructed such that after passing through specific sets of paths and time delays, these filtered random signals sum up coherently as they arrive at the target focusing points. On the other hand, they yield incoherent signals at locations with different sets of signal propagation paths. This is an extension of our previous work [10], in which we chopped the intended message signals to generate audio chunks which, in place of white Gaussian noise, were filtered and emitted by loudspeakers. After writing [10], we realized that using the chopped signals is not essential and that the results can in fact be improved by simply using white noise.

In our second approach, the idea is to send random noise from loudspeakers in addition to message signals, such that the noise signals add up to zero only at the intended listening points while they continue to mask the messages everywhere else. This results in the interception of clean audio messages at the focusing spots while having low intelligibility at other locations. This technique is inspired by standard methods in wireless networking on jamming eavesdroppers [11, 12]. However, to the best of our knowledge, the prior works consider fading wireless channels without explicitly considering inter-symbol interference (echoes). While this could be a fair assumption for networks like WiFi where sampling times are much larger than propagation delays of wireless signals, this is not the case in room acoustics. Hence, we adapt this jamming scheme to work with long convolutional channels.

Privacy in multizone reproduction systems was first studied in [13] where the authors also use noise to mask message signals in “quiet” zones to reduce intelligibility. While their method is applicable both in anechoic and in reverberant conditions, the performance is degraded in the presence of echoes. On the other hand, as we elaborate later, our methods critically rely on echoes and multipath propagation. In particular, our solutions exploit the spatial diversity of room impulse responses (RIRs) across different locations in a room, and the redundant degrees of freedom in signal transmission provided by multiple loudspeakers. Unlike in multizone methods, however, we can only deliver messages to a small, fixed region of space. On the other hand, we achieve good performance using a rather small number of loudspeakers and impulse response measurements (in our experiments we use only six).

The problem of jamming eavesdroppers has been studied extensively in wireless communication. The theoretical foundation was laid by Shannon [14] and later extended by [15, 16] who showed the feasibility of secrecy if the communication channel of an eavesdropper is degraded. The methods in [11, 12, 17] use artificial noise; [18] showed the possibility of secret communication as a consequence of slow wireless fading. Prior works have also looked at a related problem of eavesdropper detection [19, 20, 21].

In this paper, we empirically show that unlike traditional multizone sound field reproduction which is usually degraded in reverberant environments [22, 23], both of our proposed approaches give excellent results in the presence of echoes since echoes enhance spatial diversity. We derive conditions needed to generate desired messages at the focusing spots, and demonstrate both numerically and in real experiments that with six speakers and the knowledge of RIRs at the intended listening points, private audio communication is effectively achievable. In addition, we compare the robustness of the two approaches to various system failures and uncertainties.
2. PROBLEM FORMULATION

Consider a system with \( L \) loudspeakers, each playing a message signal to \( K \) listeners. Without loss of generality, let the desired length of the signal \( y_k \) at the \( k^{th} \) listener be \( N \). We also assume that the room impulse response (RIR) between the \( k^{th} \) listener and the \( i^{th} \) speaker is a sequence \( h_{ki} \) which is \( L_h \) long.

This signal received by the \( k^{th} \) listener is given as a sum of convolutions:

\[
y_k(n) = \sum_{i=1}^{L} (h_{ki} * x_i)(n), \quad n = 0, 1, ..., N - 1,
\]

where \( x_i \in \mathbb{R}^{L_x} \) is the signal transmitted by the \( i^{th} \) speaker with length \( L_x = N - L_h + 1 \). We define intended message vector \( y_{in} \in \mathbb{R}^N \), \( y_{in} = [y_1, y_2, ..., y_K] \).

Similarly, we define channel matrices \( H_k \) of size \( N \times L(L_x) \) as \( [H_{k1}, H_{k2}, ..., H_{kK}] \), where each \( H_{ki} \) is a Toeplitz convolution matrix composed using \( h_{ki} \). Defining \( H^T = [H_1^T, H_2^T, ..., H_K^T] \) and \( x^T = [x_1^T, x_2^T, ..., x_K^T] \), (1) can be rewritten as:

\[
y_{in} = Hx.
\]

If the matrix \( H \) has full row rank, we can reconstruct any desired message signals at the \( K \) listeners. A well-known solution to this is given by \( x = H^T y_{in} \), where \( H^T \) is the pseudo inverse of \( H \). Though this solution suffices for message reconstruction at the listeners, it does not enforce unintelligibility at other locations. We could, however, exploit the additional degrees of freedom provided by the nullspace of \( H \) to generate a suitable \( x \) that ensures signal degradation outside the target focusing spots.

We note that for typical audio sampling rates, RIR lengths and message lengths, \( H \) is far too large to compute the pseudoinverse explicitly. That is why we solve all least-squares design problems by the nullspace of \( H \), i.e., \( w = P_{N(H)} v \), where the entries of \( v \) are iid standard Gaussian and \( P_{N(H)} \) is the projector on the null space of \( H \).

As mentioned in Section 2, \( H \) is typically large, which makes the direct computation of its nullspace a prohibitive complex task. Instead, we first find the projection of \( v \) on the row space of \( H \) by solving

\[
\hat{z} = \arg\min_{\tilde{z}} \| v - H^T \tilde{z} \|^2_2.
\]

4. CONDITIONS FOR PERFECT RECONSTRUCTION

In this section, we present the conditions needed to ensure perfect reconstruction of any set of message signals of length \( N \) at the \( K \) listeners (or any \( y_{in} \in \mathbb{R}^{NK} \) ) for both approaches.

3. THE TWO APPROACHES

As per (2), \( x \) can be suitably chosen to ensure that the message signals outside the focusing spots remain unintelligible. In this section, we present two methods to achieve this task, each constructing \( x \) in a different way: (i) multichannel convolutional synthesis by noise and (ii) noise in the nullspace approach.

3.1. Multichannel convolutional synthesis by noise

Recall from (1) that the signal arriving at each listener is \( y_k = \sum_{L_h} h_{ki} * x_i \). In this first approach, we constrain \( x_i \) to be a convolution of a filter \( g_i \) of length \( L_g \) with a signal \( n_i \) of length \( L_n \), drawn from standard normal distribution. This is equivalent to

\[
x_i = N_i g_i, \quad i = 1, 2, ..., L.
\]

where \( N_i \) is an \( L \times L_g \) Toeplitz convolution matrix composed using the vector \( n_i \), with \( L_g = L_n + L_x - 1 \). We define \( g^T = [g_1^T, g_2^T, ..., g_L^T] \) and a block diagonal matrix \( N \) as

\[
N = \text{diag}([N_1, N_2, ..., N_L]).
\]

Then equations in (3) can be combined for all \( i \in \{1, ..., L\} \) to give

\[
x = Ng, \quad y_{in} = HNg.
\]
Intended listener 1
Intended listener 2
Other location
STOI
MCCS approach... Listeners
(c) (d)
Speakers
MCCS approach: Reverberant case
Null space approach: Reverberant case
S1 S2 S3 S4 S5 S6

that the nullspace of \( H \) that \( \text{rank}(H) \leq N \).

On the other hand, because the nullspace of \( H \) is continuously distributed and independent from \( N \), it will intersect the range of \( N \) exactly along a subspace of dimension \( L + L_g - N \) with probability one.

This result implies that for most setups in sufficiently reverberant rooms, we will be able to produce the desired messages at the listener positions.

4.2. Noise in nullspace approach

From (5), \( H \) needs to have full row rank for perfect reconstruction of all \( y_{\text{in}} \in \mathbb{R}^{NK} \). Similarly to the previous case, since \( H \) is a function of the RIRs between the speaker-listener pairs, it is not completely in the user’s control to ensure that it has full rank as it depends on geometry and spatial diversity of RIRs. In practice, if we assume a randomized setup and room as in the previous section, \( H \) can be expected to have full row rank with probability one.

**Proposition 4.2.** The following conditions are necessary for \( H \) to have full row rank.

(a) The number of rows of \( H \) should be at least as large as the length of \( y_{\text{in}} \), i.e., \( L_x + L_h - 1 \geq N \).

(b) There should be at least as many columns as rows in \( H \).

(c) \( L_x \) needs to be greater than the highest relative delay among each listener-speaker pair.

**Proof.** (a) ensures that we have sufficient samples to generate the desired message length; (b) is elementary linear algebra; (c) ensures that “silent” regions do not exist within a signal generated at a listening point.

It should be noted that (b) gives a lower bound on the number of listeners, \( L \), needed for reconstruction, i.e., \( L \geq \frac{K (L_x + L_h - 1)}{L_x} \), which is lower than the number of speakers needed by the MCCS approach, as per Proposition 4.1.

5. EXPERIMENTAL RESULTS

We evaluate the performance of the two proposed techniques using both numerical and real experiments. The numerical experiments are performed with 6 loudspeakers randomly placed in a convex simulated room of size 7 m × 8 m having walls with absorption coefficient 0.35. RIRs between the speakers and listeners are calculated based on image source model, using the pyroomacoustics package [24]. We perform the real experiments in an office space of size 10 m × 6 m using two Genelec 8030B and four Genelec 8010A loudspeakers. The RIRs are measured using the exponential sweep technique [25]. In all experiments, the power of signals emitted by the loudspeakers is kept fixed. The intelligibility of the generated sounds is assessed using Short-Time Objective Intelligibility (STOI) [26] measure.

5.1. Numerical experiments

5.1.1. Perfect reconstruction: A case for echoes

In order to provide an insight into the importance of echoes in our solution, we first perform an experiment in a simulated anechoic room. We randomly place two listeners inside the room and calculate STOI values of the signals arriving there using the two approaches. An additional location is randomly chosen to check how degraded the audio signals appear outside the target focusing spots. We then repeat the same experiment but in the presence of echoes. Fig. 1(a) shows that in the anechoic setting, while the signal at the first listener has a high intelligibility with STOI values close to 1 for both approaches, the second listener does not. On the other hand, Fig. 1(b) shows that in the presence of echoes, signal intelligibility is restored at the second listener as well. This indicates that the spatial diversity provided by echoes helps in conditioning the channel matrix \( H \), which in turn supports perfect reconstruction of messages at target locations.

5.1.2. Signal degradation outside focusing spots

Both Fig. 1(a) and (b) indicate that the nullspace-based method has a greater impact on signal degradation at the location chosen outside the focusing spots. To examine this further, we calculate STOI scores at 4200 locations in a simulated reverberant room and create maps as shown in Fig. 1(c) and (d). The bright spots at the locations of focusing points indicate regions of high intelligibility in both plots, whereas the relatively dark regions in Fig. 1(d) represent lower STOI values for the nullspace approach and, thus, reduced intelligibility as compared to the MCCS approach in Fig. 1(c).

Both methods perform signal degradation outside the focusing spots using noise vectors. To understand how these random vectors result in unintelligibility of sound, we first investigate the role of noise variance. For 100 randomly selected speaker-listener configurations, we check the impact of increasing noise variance on STOI values. Fig. 2(a) shows a decline in median STOI scores as the input noise power is increased for the nullspace approach, whereas they do not change much for the MCCS method.
This result is not surprising because in the nullspace approach, noise is fed into the loudspeakers with the message signals in an additive sense. Thus, a deterioration of SNR and subsequent STOI hit is expected with increase in noise variance. On the other hand, from Section 3.1, the signal emitted by the $i^{th}$ loudspeaker is $x_i = n_i + g_i$. In this setting, if the variance of each $n_i$ sample is increased, the filter $g_i$ is simply scaled to preserve the original $x_i$.

We now investigate the factors that impact the jamming capability of the MCCS approach. Recall that this method involves “scrambling” of message-carrying input filters $g_i$ by noise which are thereby appropriately descrambled at the intended locations by the correct RIR values. Thus, we expect that longer noise vectors would have a stronger impact on signal integrity when the RIR changes. To verify this claim, we vary the length of noise vectors $L_n$ as a proportion of a fixed length $L_x$, and calculate the STOI scores for 100 randomly chosen speaker-listener configurations. Fig. 2(b) verifies that increasing the length of noise vectors leads to a decrease in median intelligibility scores outside the focusing spots.

These results point towards an interesting phenomenon. For the nullspace approach, the jamming capability can be improved by increasing the input noise power which is upper bounded by the input power constraints at the loudspeaker. On the other hand, in MCCS approach, for a fixed message length $N$ and fixed $L_n$, $L_x = L_n + L_n - 1$ is fixed. Thus, jamming can be improved by increasing $L_n$ as long as $L_n \geq \frac{N}{N+1}$ (from Proposition 4.1).

5.1.3. Robustness to system uncertainties

Here, we assess how the reconstruction of audio messages at the target listeners is affected due to system uncertainties: in particular, the impact of malfunction of a set of speakers after the appropriate $x_i$ have been estimated, and inaccuracies in the measurement of RIR values. For this, we did simulations over 100 random speaker listener configurations, and checked how the STOI scores were affected. Fig. 3(a) indicates that the STOI values for MCCS method decline less rapidly with increasing speaker drops as compared to the nullspace method. Similarly, Fig. 3(b) indicates that errors in the knowledge of RIRs before signal transmission by the loudspeakers lead to reduced intelligibility in the focusing spots. Again, the MCCS approach shows more robustness to system errors as compared to the nullspace approach.

6. CONCLUSION

In this paper, we presented two approaches to address private audio communication problem in a reverberant room. Both approaches are based on emitting noise signals from loudspeakers and then utilizing the echoes in the room to ensure that they yield intelligible messages at selected locations, while being incoherent elsewhere. Simulated and real experiments suggest that with just six loudspeakers and a few impulse response measurements, we can deliver clear audio messages at the desired locations while ensuring unintelligibility everywhere else. The experiments further suggest that the nullspace based method is more capable of jamming locations outside the targeted focusing spots, whereas the MCCS method is more robust to errors in system design.
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