

# Multichannel Signal Processing With Deep Neural Networks for Automatic Speech Recognition

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**Abstract**—Multichannel automatic speech recognition (ASR) systems commonly separate speech enhancement, including localization, beamforming, and postfiltering, from acoustic modeling. In this paper, we perform multichannel enhancement jointly with acoustic modeling in a deep neural network framework. Inspired by beamforming, which leverages differences in the fine time structure of the signal at different microphones to filter energy arriving from different directions, we explore modeling the raw time-domain waveform directly. We introduce a neural network architecture, which performs multichannel filtering in the first layer of the network, and show that this network learns to be robust to varying target speaker direction of arrival, performing as well as a model that is given oracle knowledge of the true target speaker direction. Next, we show how performance can be improved by *factoring* the first layer to separate the multichannel spatial filtering operation from a single channel filterbank which computes a frequency decomposition. We also introduce an adaptive variant, which updates the spatial filter coefficients at each time frame based on the previous inputs. Finally, we demonstrate that these approaches can be implemented more efficiently in the frequency domain. Overall, we find that such multichannel neural networks give a relative word error rate improvement of more than 5% compared to a traditional beamforming-based multichannel ASR system and more than 10% compared to a single channel waveform model.

**Index Terms**—Beamforming, deep learning, noise-robust speech recognition.

## I. INTRODUCTION

WHILE state-of-the-art automatic speech recognition (ASR) systems perform reasonably well in close-talking microphone conditions, performance degrades in conditions when the microphone is far from the user. In such farfield

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cases, the speech signal is degraded by reverberation and additive noise. To improve recognition in such cases, ASR systems often use signals from multiple microphones to enhance the speech signal and reduce the impact of reverberation and noise [1]–[3].

Multichannel ASR systems often use separate modules to perform recognition. First, microphone array speech enhancement is applied, typically broken into localization, beamforming and postfiltering stages. The resulting single channel enhanced signal is passed to an conventional acoustic model [4], [5]. A commonly used enhancement technique is filter-and-sum beamforming [2], which begins by aligning signals from different microphones in time (via localization) to adjust for the propagation delay from the target speaker to each microphone. The time-aligned signals are then passed through a filter (different for each microphone) and summed to enhance the signal from the target direction and to attenuate noise coming from other directions. Commonly used filter design criteria are based on Minimum Variance Distortionless Response (MVDR) [3], [6] or multichannel Wiener filtering (MWF) [1].

When the end goal is to improve ASR performance, tuning the enhancement model independently from the acoustic model might not be optimal. In an early effort to address this issue [7] proposed a likelihood-maximizing beamforming (LIMABEAM) which optimizes beamformer parameters jointly with those of the acoustic model. This technique was shown to outperform conventional techniques such as delay-and-sum beamforming (i.e. filter-and-sum where the filters consist of impulses). Like most enhancement techniques, LIMABEAM is a model-based scheme and requires an iterative algorithm that alternates between acoustic model inference and enhancement model parameter optimization. Contemporary acoustic models are generally based on neural networks, optimized using a gradient learning algorithm. Combining model-based enhancement with an acoustic model that uses gradient learning can lead to considerable complexity, e.g. [8].

In this paper we extend the idea of performing beamforming jointly with acoustic modeling from [7], but do this within the context of a deep neural network (DNN) framework by training an acoustic model directly on the raw signal. DNNs are attractive because they have been shown to be able to perform feature extraction jointly with classification [9]. Previous work

has demonstrated the possibility of training deep networks directly on raw, single channel, time domain waveform samples [10]–[15]. The goal of this paper is to explore a variety of different joint enhancement/acoustic modeling DNN architectures that operate on multichannel signals. We will show that jointly optimizing both stages is more effective than techniques which cascade independently tuned enhancement algorithms with acoustic models.

Since beamforming takes advantage of the fine time structure of the signal at different microphones, we begin by modeling the raw time-domain waveform directly. In this model, introduced in [14], [16], the first layer consists of multiple time convolution filters, which map the multiple microphone signals down to a single time-frequency representation. As we will show, this layer learns bandpass filters which are spatially selective, often learning several filters with nearly identical frequency response, but with nulls steered toward different directions of arrival. The output of this spectral filtering layer is passed to an acoustic model, such as a convolutional long short-term memory, deep neural network (CLDNN) acoustic model [17]. All layers of the network are trained jointly.

As described above, it is common for multichannel speech recognition systems to perform spatial filtering independently from single channel feature extraction. With this in mind, we next investigate explicitly factorizing these two operations to be separate neural network layers. The first layer in this “factored” raw waveform model consists of short-duration multichannel time convolution filters which map multichannel inputs down to a single channel, with the idea that the network might learn to perform broadband spatial filtering in this layer. By learning several filters in this “spatial filtering layer”, we hypothesize that the network can learn filters for multiple different spatial look directions. The single channel waveform output of each spatial filter is passed to a longer-duration time convolution “spectral filtering layer” intended to perform finer frequency resolution spectral decomposition analogous to a time-domain auditory filterbank as in [15]. The output of this spectral filtering layer is also passed to an acoustic model, and all layers of the network are trained jointly.

One issue with the two architectures above is that once weights are learned during training, they remain fixed for each test utterance. In contrast, some beamforming techniques, such as the generalized sidelobe canceller [18], update weights adaptively within each utterance. We explore an adaptive neural net architecture, where an LSTM is used to predict spatial filter coefficients that are updated at each frame. These filters are used to filter and sum the multichannel input, replacing the “spatial filtering layer” of the factored model described above, before passing the enhanced single channel output to a waveform acoustic model.

Finally, since convolution between two time domain signals is equivalent to the element-wise product of their frequency domain counterparts, we investigate speeding up the raw waveform neural network architectures described above by consuming the complex-valued fast Fourier transform of the raw input and implementing filters in the frequency domain.

## II. EXPERIMENTAL SETUP

### A. Data

We conduct experiments on a dataset comprised of about 2,000 hours of noisy training data consisting of 3 million English utterances. This data set is created by artificially corrupting clean utterances using a room simulator to add varying degrees of noise and reverberation. The clean utterances are anonymized and hand-transcribed voice search queries, and are representative of Google’s voice search traffic. Noise signals, which include music and ambient noise sampled from YouTube and recordings of “daily life” environments, are added to the clean utterances at SNRs ranging from 0 to 20 dB, with an average of about 12 dB. Reverberation is simulated using the image method [19] – room dimensions and microphone array positions are randomly sampled from 100 possible room configurations with  $T_{60}$  s ranging from 400 to 900 ms, with an average of about 600 ms. The simulation uses an 8-channel uniform linear microphone array, with inter-microphone spacing of 2 cm. Both noise source location and target speaker locations change between utterances; the distance between the sound source and the microphone array varies between 1 to 4 meters.

The primary evaluation set consists of a separate set of about 30,000 utterances (over 20 hours), and is created by simulating similar SNR and reverberation settings to the training set. Care was taken to ensure that the room configurations, SNR values,  $T_{60}$  times, and target speaker and noise positions in the evaluation set differ from those in the training set. The microphone array geometry between the training and simulated test sets is identical. Most of the results we report will be on this test set.

We obtained a second “rerecorded” test set by playing the evaluation set and the noises separately using a mouth simulator and a speaker, respectively, in a living room setting. The signals are recorded using a 7-channel circular microphony array with a radius of 3.75 cm. Assuming an x-axis that passes through two diagonally opposite mics along the circumference of the array, the angle of arrival of the target speaker ranges from 0 to 180 degrees. Noise originates from locations different from the target speaker. The distance of the sources to the target ranges from 1 to 6 meters. To create noisy rerecorded eval sets, the rerecorded speech and noise signals are mixed artificially after scaling noise to obtain SNRs ranging from 0 to 20 dB. The distribution of the SNR matches the distribution used to generate the simulated evaluation set. The average  $T_{60}$  for this set is around 200ms. We create 4 versions of the rerecorded sets to measure generalization performance of our models. The first two have rerecorded speech w/o any added noise. The mic array is placed at the center of the room and closer to the wall, respectively, to capture reasonably different reverberation characteristics. The remaining two subsets correspond to the noisy versions of these sets.

### B. Baseline Acoustic Model

We compare the models proposed in this paper to a baseline CLDNN acoustic model trained using log-mel features [17]

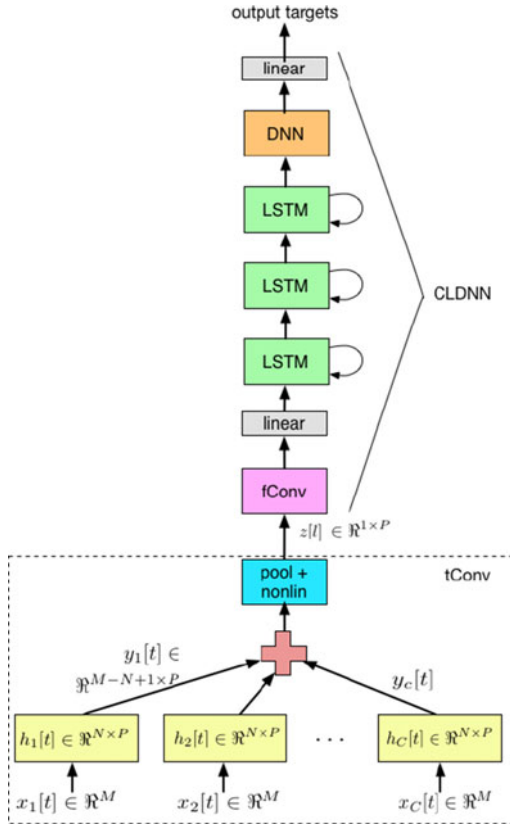


Fig. 1. Multichannel raw waveform CLDNN architecture.

computed with a 25 ms window and a 10 ms hop. Single channel models are trained using signals from channel 1,  $C = 2$  channel models use channels 1 and 8 (14 cm spacing),  $C = 4$  channel models use channels 1, 3, 6, and 8 (14 cm array span, with adjacent microphone spacing of 4 cm-6 cm-4 cm).

The baseline CLDNN architecture is shown in the CLDNN module of Fig. 1. First, the fConv layer performs convolution across the frequency dimension of the input log-mel time-frequency feature to gain some invariance to pitch and vocal tract length. The architecture used for this convolutional layer is similar to that proposed in [20]. Specifically, a single convolutional layer with 256 filters of size  $1 \times 8$  in time-frequency is used. Our pooling strategy is to use non-overlapping max pooling along the frequency axis, with a pooling size of 3. The pooled output is given to a 256-dimensional linear low-rank layer.

The output of frequency convolution is passed to a stack of LSTM layers, which model the signal across long time scales. We use 3 LSTM layers, each comprised of 832 cells, and a 512 unit projection layer for dimensionality reduction following [21]. Finally, we pass the final LSTM output to a single fully connected DNN layer comprised of 1,024 hidden units. Due to the high dimensionality of the 13,522 context-dependent state output targets used by the language model, a 512-dimensional linear output low rank projection layer is used prior to the softmax layer to reduce the number of parameters in the overall model [22]. Some experiments in the paper do not use the frequency convolution layer, and we will refer to such acoustic

models as LDNNs. It is important to note that all methods presented in this paper use an LSTM-type architecture, which has been shown to work much better with larger amounts of data [15] and is much more well-suited for our task.

During training, the CLDNN is unrolled for 20 time steps and trained using truncated backpropagation through time (BPTT). In addition, the output state label is delayed by 5 frames, as we have observed that information about future frames helps to better predict the current frame [17].

All neural networks are trained using asynchronous stochastic gradient descent (ASGD) optimization [23] to optimize a cross-entropy (CE) criterion. Unless otherwise indicated, all word error rate (WER) numbers in this paper are presented with the CE criterion. Additional sequence training experiments also use distributed ASGD [24]. All networks have 13,522 context-dependent (CD) output targets. The weights for all CNN and DNN layers are initialized using the Glorot-Bengio strategy [25], while those of all LSTM layers are randomly initialized using a uniform distribution between  $-0.02$  and  $0.02$ . We use an exponentially decaying learning rate, initialized to 0.004 and decaying by 0.1 over 15 billion frames.

### III. MULTICHANNEL RAW WAVEFORM NEURAL NETWORK

#### A. Motivation

The proposed multichannel raw waveform CLDNN is related to filter-and-sum beamforming, a generalization of delay-and-sum beamforming which filters the signal from each microphone using a finite impulse response (FIR) filter before summing them. Using similar notation to [7], filter-and-sum enhancement can be written as follows:

$$y[t] = \sum_{c=0}^{C-1} \sum_{n=0}^{N-1} h_c[n] x_c[t - n - \tau_c] \quad (1)$$

where  $h_c[n]$  is the  $n$ th tap of the filter associated with microphone  $c$ ,  $x_c[t]$ , is the signal received by microphone  $c$  at time  $t$ ,  $\tau_c$  is the steering time difference of arrival induced in the signal received by microphone  $c$  used to align it to the other array channels, and  $y[t]$  is the output signal.  $C$  is the number of microphones in the array and  $N$  is the number of FIR filter taps.

#### B. Multichannel Filtering in the Time Domain

Enhancement algorithms implementing (1) generally depend on an estimate of the steering delay  $\tau_c$  obtained using a separate localization model, and they compute filter parameters  $h_c[n]$  by optimizing an objective such as MVDR. In contrast, our aim is to allow the network to jointly estimate steering delays and filter parameters by optimizing an acoustic modeling classification objective. The model captures different steering delays using a bank of  $P$  multichannel filters. The output of filter  $p \in \{0, \dots, P-1\}$  can be written as follows:

$$y^p[t] = \sum_{c=0}^{C-1} \sum_{n=0}^{N-1} h_c^p[n] x_c[t - n] = \sum_{c=0}^{C-1} x_c[t] * h_c^p \quad (2)$$



where the steering delays are implicitly absorbed into the filter parameters  $h_c^p[n]$ . In this equation, ‘\*’ denotes the convolution operation.

The first layer in our raw waveform architecture [26] implements (2) as a multichannel convolution (in time) with a FIR spatial filterbank  $h_c = \{h_c^1, h_c^2, \dots, h_c^P\}$  where  $h_c \in \mathbb{R}^{N \times P}$  for  $c \in 0, \dots, C-1$ . Each filter  $h_c^p$  is convolved with the corresponding input channel  $x_c$ , and the overall output for filter  $p$  is computed by summing the result of this convolution across all channels  $c \in \{0, \dots, C-1\}$ . The operation within each filter is equivalent to an FIR filter-and-sum beamformer, except that it does not explicitly shift the signal in each channel by an estimated time difference of arrival. As we will show, the network learns parameters for a fixed set of filters that give good speech recognition performance.

The output signal remains at the same sampling rate as the input signal, which contains more information than is typically relevant for acoustic modeling. In order to produce an output that is invariant to perceptually and semantically identical sounds appearing at different time shifts we pool the outputs in time after filtering [14], [15], in an operation that has an effect similar to discarding the phase in the short-time Fourier transform. Specifically, the output of the filterbank is max-pooled across time to give a degree of short term shift invariance, and then passed through a compressive non-linearity.

As shown in [14], [15], single channel time convolution layers similar to the one described above implement a conventional time-domain filterbank. Such layers are capable of implementing, for example, a standard gammatone filterbank, which consists of a bank of time-domain filters followed by rectification and averaging over a small window. Given sufficiently large  $P$ , the corresponding multichannel layer can (and as we will show, does in fact) similarly implement a frequency decomposition in addition to spatial filtering. We will therefore subsequently refer to the output of this layer as a “time-frequency” feature representation.

A schematic of the multichannel time convolution layer is shown in the tConv block of Fig. 1. First, we take a small window of the raw waveform of length  $M$  samples for each channel  $C$ , denoted as  $\{x_0[t], x_1[t], \dots, x_{C-1}[t]\}$  for  $t \in 1, \dots, M$ . The signal from each channel  $x_c$  is convolved with a bank of  $P$  filters with  $N$  taps  $h_c[n] = \{h_c^1[n], h_c^2[n], \dots, h_c^P[n]\}$ . When the convolution is strided by 1 in time across  $M$  samples, the output from the convolution in each channel is  $y_c[t] \in \mathbb{R}^{(M-N+1) \times P}$ . After summing  $y_c[t]$  across channels  $c$ , we max pool the filterbank output in time (thereby discarding short term phase information), over the entire time length of the output signal  $M - N + 1$ , to produce  $y[t] \in \mathbb{R}^{1 \times P}$ . Finally, we apply a rectified non-linearity, followed by a stabilized logarithm compression<sup>1</sup>, to produce  $z[l]$ , a  $P$  dimensional frame-level feature vector at frame  $l$ . We then shift the window around the waveform by 10 ms and repeat this time convolution, producing a sequence of feature frames at 10 ms intervals.

<sup>1</sup>We use a small additive offset to truncate the output range and avoid numerical instability with very small inputs:  $\log(\cdot + 0.01)$ .

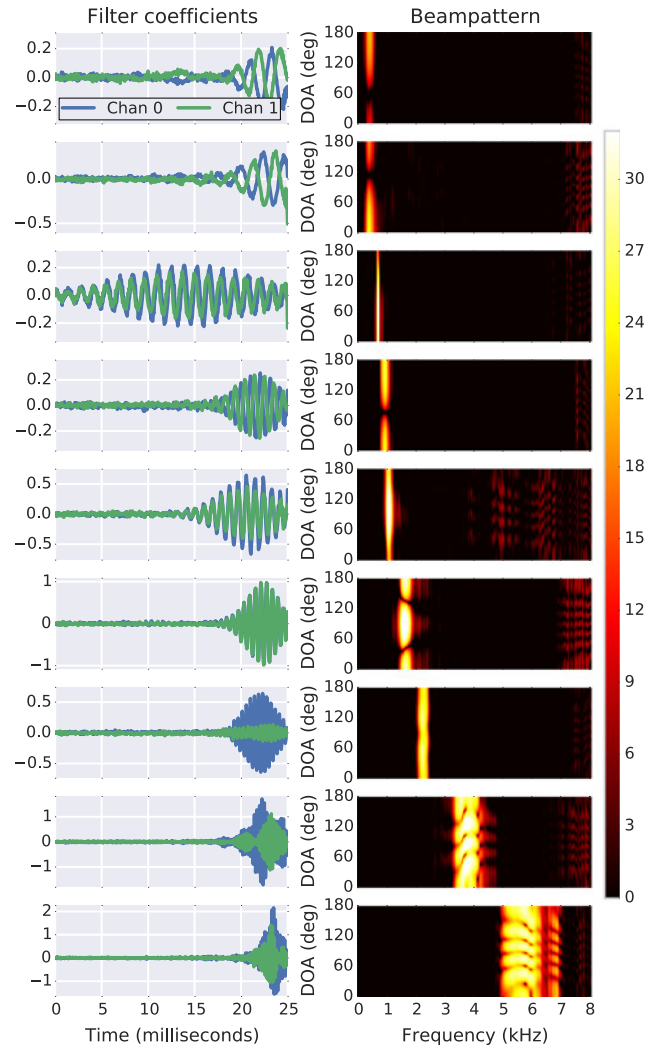


Fig. 2. Example filter coefficients and corresponding spatial response beampatterns learned in a network with 128 tConv filters trained on 2 channel inputs. Some filters learned by this network have nearly-identical center frequencies but different spatial responses. For example, the top two example filters both have center frequencies of about 440 Hz, but the first filter has a null at a direction of arrival of about 60 degrees, while the second has a null at about 120 degrees. The corresponding phase difference between the two channels of each filter is visible in the time domain filter coefficients plotted on the left.

To match the time-scale of the log-mel features, the raw waveform features are computed with an identical filter size of 25 ms, or  $N = 400$  at a sampling rate of 16 kHz. The input window size is 35 ms ( $M = 560$ ) giving a 10 ms fully overlapping pooling window. Our experiments explore varying the number of time-convolutional filters  $P$ .

As shown in the CLDNN block of Fig. 1, the output of the time convolutional layer (tConv) produces a frame-level feature, denoted as  $z[l] \in \mathbb{R}^{1 \times P}$ . This feature is then passed to a CLDNN model [17] described in Section II, which predicts context dependent state output targets.

### C. Filterbank Spatial Diversity

Fig. 2 plots example multichannel filter coefficients and their corresponding spatial responses, or beampatterns, after train-

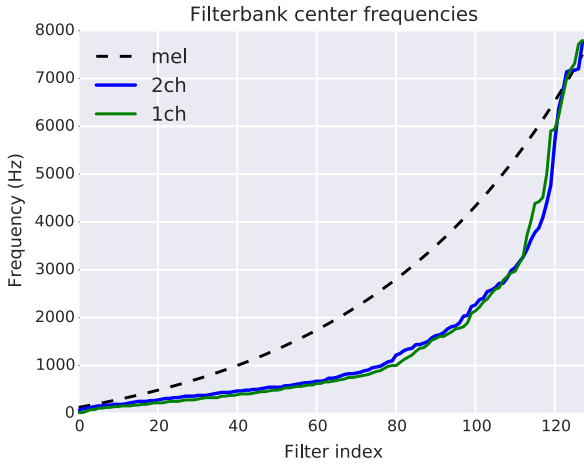


Fig. 3. Comparison of the peak response frequencies of waveform CLDNN filterbanks trained on one- and two-channel inputs to the standard mel frequency scale.

ing for  $\tau_{\text{Conv}}$ . The beampatterns show the magnitude response in dB as a function of frequency and direction of arrival, i.e. each horizontal slice of the beampattern corresponds to the filter’s magnitude response for a signal coming from a particular direction, and each vertical slice corresponds to the filter’s response across all spatial directions in a particular frequency band. Lighter shades indicate regions of the frequency-directions space which are passed through the filter, while darker shades indicate regions which are filtered out. Within a given beampattern, we refer to the frequency band containing the maximum overall response as the filter’s *center frequency* (since the filters are primarily bandpass in frequency), and the direction corresponding to the minimum response in that frequency as the filter’s *null direction*.

The network tends to learn filter coefficients with very similar shapes in each channel except they are slightly shifted relative to each other, consistent with the notion of a steering delay  $\tau_c$  described in Section III. Most filters have a bandpass response in frequency, with bandwidths that increase with center frequency, and many are steered to have stronger response for signals arriving from a particular direction. Approximately two-thirds of the filters in the model shown in Fig. 2 demonstrate a significant spatial response, i.e. show a difference of at least 6 dB between the direction with the minimum and maximum response at the filter center frequency. Such strong spatial responses are clearly visible in the null near 120 degrees in the second filter, and a similar null near 60 degrees in the fourth filter shown in Fig. 2.

Fig. 3 plots the peak response frequencies of filterbanks from networks trained on a one- and two-channel networks of the form shown in 2. The two networks converge to similar frequency scales, both consistently allocating many more filters to low frequencies compared to the mel scale. For example, the learned filterbanks have roughly 80 filters with peak responses below 1000 Hz, while a 128-band mel scale has only 40 bands with center frequencies below 1000 Hz. The network furthermore tends to learn subsets of filters with the same overall shape and frequency response but tuned to have nulls in different

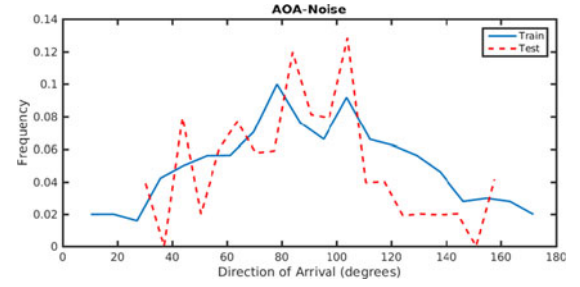


Fig. 4. Distribution of the direction of arrival of noise in training and test set.

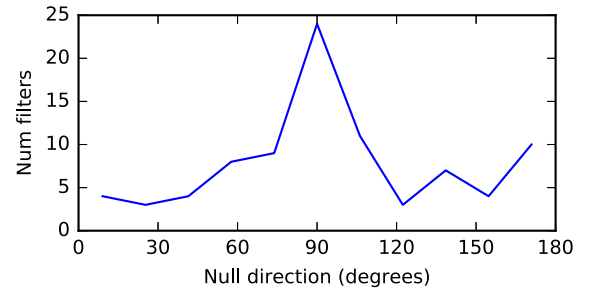


Fig. 5. Histogram of null directions of spatial filters from a 2 channel model with 128 filters. “Null direction” is computed as the direction of minimum response for filters where the minimum response is at least 6 dB below the maximum. (Filters for which minimum and maximum directional responses differ by less than 6 dB are not included in the plot.)

TABLE I  
WER FOR RAW WAVEFORM MULTICHANNEL CLDNNs WITH DIFFERENT  
NUMBER OF INPUT CHANNELS

Filters	2ch (14 cm)	4ch (4-6-4 cm)	8ch (2 cm)
128	21.8	21.3	21.1
256	21.7	20.8	20.6
512	-	20.8	20.6

The inter-microphone spacing is given in parentheses.

directions, as illustrated by the top two example filters in Fig. 2. Such diversity in spatial response gives upstream layers information that can be used to discriminate between signals arriving from different directions.

The ability of the network to exploit directional cues is constrained by the number of filters it uses. By increasing the number of filters, we can potentially improve the spatial diversity of the learned filters and therefore allow the network to better exploit directional cues.

To see how the distribution of null directions learned by the filters corresponds to the direction of arrival (DOA) of noise, Fig. 4 plots the distribution of noise DOA for the training and test sets. Notice there is a strong correlation between the noise DOA distribution and the learned filter null directions in Fig. 5, illustrating that the learned filters are learning to filter out noise.

Table I demonstrates the effect of increasing the number of filters on overall word error rate (WER). Improvements saturate at 128 filters for networks trained on 2 channel inputs, while 4 and 8 channels networks continue to improve with 256 filters.

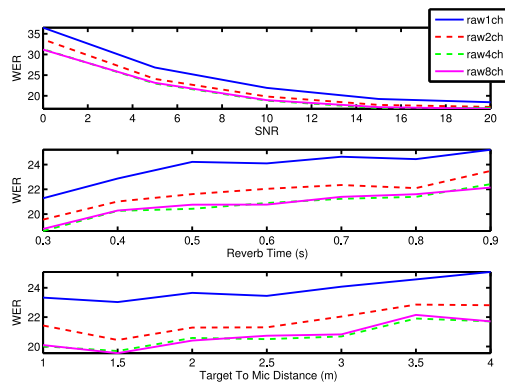


Fig. 6. WER breakdown for multichannel models.

TABLE II  
WER FOR LOG-MEL MULTICHANNEL CLDNNs

Filters	2ch (14 cm)	4ch (4-6-4 cm)	8ch (2 cm)
128	22.0	21.7	22.0
256	21.8	21.6	21.7

With additional input channels the  $\text{tConv}$  filters are able to learn more complex spatial responses (even though the total array span is unchanged), enabling the network to make use of additional filterbank capacity to improve performance.

Multi-microphone signal processing can help to enhance the signal and suppress noise. Therefore, we should expect to see improvements in WER as the number of microphones increases, especially under more challenging conditions. A breakdown of WER in Fig. 6 shows that this is indeed the case – the performance improvement of the 8 channel system over the 1 and 2 channel systems is largest in low SNR and high reverberation time. Notice also that there is very little difference in performance going from 4 to 8 channels.

#### D. Comparison to Log-Mel

We train baseline multichannel log-mel CLDNNs by computing log-mel features for each channel, and treating these as separate feature maps into the CLDNN. Since the raw waveform model improves as we increase the number of filters, we perform the same experiment for log-mel. Table II shows that for log-mel, neither increasing the number of filters (frequency bands) nor increasing the number of microphone channels has a strong effect on word error rate. Since log-mel features are computed from the FFT magnitude, the fine time structure (stored in the phase), and therefore information about inter-microphone delays, is discarded. Log-mel models can therefore only make use of the weaker inter-microphone level difference cues. However, the multichannel time-domain filterbanks in the raw waveform models utilize the fine time structure and show larger improvements as the number of filters increase.

Comparing Tables I and II we can see that raw waveform models consistently outperform log-mel, particularly for larger number of channels where more spatial diversity is possible.

TABLE III  
WER WITH ORACLE KNOWLEDGE OF THE TRUE TARGET TDOA

Feature	1ch	2ch (14 cm)	4ch (4-6-4 cm)	8ch (2 cm)
oracle D+S	23.5	22.8	22.5	22.4
oracle TAM	23.5	21.7	21.3	21.3
raw, no tdoa	23.5	21.8	21.3	21.1

All models use 128 filters.

#### E. Comparison to Oracle Knowledge of Speech TDOA

Note that the models presented in the previous subsection do not explicitly estimate the time delay of arrival of the target source arriving at different microphones, which is commonly done in beamforming [2]. Time delay of arrival (TDOA) estimation is useful because time aligning and combining signals steers the array such that the target speech signal is enhanced relative to noise sources coming from other directions.

In this section, we analyze the behavior of raw waveform CLDNNs when the signals are time aligned using the true TDOA calculated using the room geometry. For the delay-and-sum (D+S) approach, we shift the signal in each channel by the corresponding TDOA, average them together, and pass the result into a 1-channel raw waveform CLDNN. For the time-aligned multichannel (TAM) approach, we align the signals in time and pass them as separate channel inputs to a multichannel raw waveform CLDNN. Thus the difference between the multichannel raw waveform CLDNNs described in Section 2 and TAM is solely in how the data is presented to the network (whether or not they are first explicitly aligned to “steer” toward the target speaker direction); the network architectures are identical.

Table III compares the WER of D+S, TAM, and raw waveform models when we do not shift the signals by the TDOA. First, notice that as we increase the number of channels, D+S continues to improve, since finer spatial sampling reduces the sidelobes of the spatial response, leading to increased suppression of noise and reverberation energy arriving from other directions. Second, notice that TAM always has better performance than D+S, as TAM is more general than D+S because it allows individual channels to be filtered before being combined. But notice that the raw waveform CLDNN, without any explicit time alignment or localization (TDOA estimation), performs as well as TAM with the time alignment. This shows us that the trained un-aligned network is implicitly robust to varying TDOA.

#### F. Summary

To conclude this section, we show the results after sequence training in Table IV. We compare to results for 8 channel oracle D+S, where the true target speech TDOA is known, and to oracle MVDR [6] where the true noise covariance is known in addition to the target TDOA. Oracle MVDR has been shown to perform well, especially on simulated spatialized data [27], [28] similar to what we use here. Note that because we are using the oracle noise covariance, our oracle MVDR does not suffer from target cancellation due to reverberation, a common failure

TABLE IV  
RAW WAVEFORM MODEL WER AFTER SEQUENCE TRAINING

Model	Filters	WER - CE	WER - Seq
raw, 1ch	128	23.5	19.3
D+S, 8ch, oracle	128	22.4	18.8
MVDR, 8ch, oracle	128	22.5	18.7
raw, unfactored, 2ch	128	21.8	18.2
raw, unfactored, 4ch	256	20.8	17.2
raw, unfactored, 8ch	256	20.6	17.2

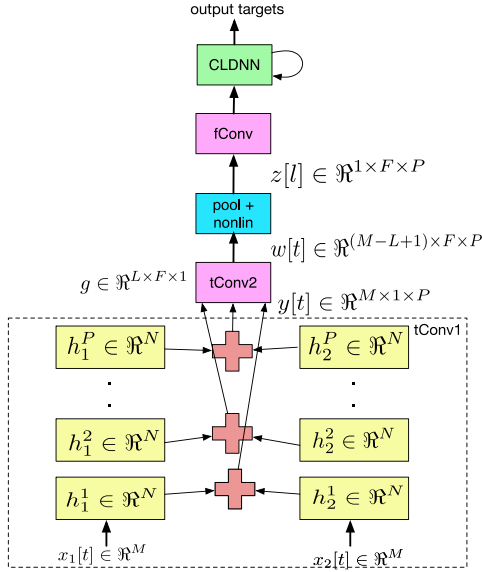


Fig. 7. Factored multichannel raw waveform CLDNN architecture for  $P$  look directions. The figure shows two channels for simplicity.

mode for non-oracle MVDR. Table IV shows that the raw unfactored model, even using only 2 channel inputs and no oracle information, outperforms the single channel and oracle signal processing methods. Using 4 channel inputs, the raw-waveform unfactored model achieves between an 8–10% relative improvement over single channel, D+S and MVDR.

#### IV. FACTORING SPATIAL AND SPECTRAL SELECTIVITY

##### A. Architecture

In multichannel speech recognition systems, multichannel spatial filtering is often performed separately from single channel feature extraction. However, in the unfactored raw-waveform model, spatial and spectral filtering are done in one layer of the network. In this section, we factor out spatial and spectral filtering into separate layers [26], as shown in Fig. 7.

The motivation for this architecture is to design the first layer to be spatially selective, while implementing a frequency decomposition shared across all spatial filters in the second layer. Thus the combined output of the second layer will be the Cartesian product of all spatial and spectral filters.

The first layer, denoted by  $\text{tConv1}$  in the figure, again models (2) and performs a multichannel time-convolution with a FIR spatial filterbank. The operation of each filter

$p \in \{0, \dots, P-1\}$ , which we will refer to as a spatial look direction in the factored model, can again be interpreted as a filter-and-sum beamformer, except that any overall time shift is implicit in the filter coefficients rather than being explicitly represented as in (1). The main differences between the unfactored and factored approaches are as follows. First, both the filter size  $N$  and number of filters  $P$  are much smaller in order to encourage the network to learn filters with a broadband response in frequency that span a small number of spatial look directions needed to cover all possible target speaker locations. The shorter filters in this layer will have worse frequency resolution than those in the unfactored model, but that will be dealt with in the next layer. We hope that this poor frequency resolution will encourage the network to use this first layer to focus on spatial filtering, with a limited spectral response. To make the combination of the first two layers of the factored model conceptually similar to the first layer of the unfactored model (i.e., a bank of bandpassed beamformers), the multi-channel (first) filter layer is not followed by any non-linear compression (i.e. ReLU, log), and we do not perform any pooling between the first and second layers.

The second time-convolution layer, denoted by  $\text{tConv2}$  in the figure, consists of longer-duration single-channel filters. It therefore can learn a decomposition with better frequency resolution than the first layer but is incapable of doing any spatial filtering. Given the  $P$  feature maps from the first layer, we perform a time convolution on each of these signals, very similar to the single-channel time-convolution layer described in [15], except that the time convolution is shared across all  $P$  feature maps or “look directions”. We denote this layer’s filters as  $g \in \mathbb{R}^{L \times F \times 1}$ , where 1 indicates sharing across the  $P$  input feature maps. The “valid” convolution produces an output  $w[t] \in \mathbb{R}^{(M-L+1) \times F \times P}$ . The output of the spectral convolution layer, for each look direction  $p$  and each filter  $f$ , is given by (3).

$$w_f^p[t] = y^p[t] * g_f \quad (3)$$

Next, we pool the filterbank output in time thereby discarding short-time (i.e. phase) information, over the entire time length of the output signal, to produce an output of dimension  $1 \times F \times P$ . Finally, we apply a rectified non-linearity, followed by a stabilized logarithm compression, to produce a frame-level feature vector at frame  $l$ , i.e.,  $z_l \in \mathbb{R}^{1 \times F \times P}$ , which is then passed to a CLDNN model. We then shift the window of the raw waveform by a small (10ms) hop and repeat this time convolution to produce a set of time-frequency-direction frames at 10ms intervals.

The output of the time convolutional layer ( $\text{tConv2}$ ) produces a frame-level feature, denoted as  $z[l] \in \mathbb{R}^{1 \times F \times P}$ , which is then passed to a CLDNN acoustic model.

##### B. Number of Spatial Filters

We first explore the behavior of the proposed factored multichannel architecture as the number of spatial filters  $P$  varies. Table V shows that we get good improvements up to 10 spatial filters. We did not explore above 10 filters due to the computational complexities of passing 10 feature maps to the  $\text{tConv2}$



TABLE V  
WER WHEN VARYING THE SIZE OF THE SPATIAL FILTERS IN  $\text{tConv1}$

# Spatial Filters $P$	WER
baseline 2 ch, raw [16]	21.8
1	23.6
3	21.6
5	20.7
10	20.4

All models use 128 filters for  $\text{tConv2}$  and results are presented for 2 channels.

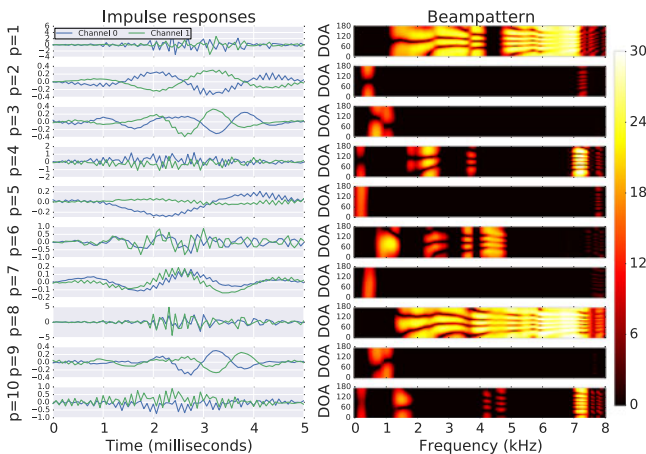


Fig. 8. Trained filters and spatial responses for 10 spatial directions.

layer. Furthermore, we later found that after sequence training, there was no difference in performance between 5 and 10 spatial filters [29].

The factored network, with 10 spatial filters, achieves a WER of 20.4%, a 6% relative improvement over the 2 channel unfactored multichannel raw-waveform CLDNN. It is important to note that since the  $\text{tConv2}$  layer is shared across all look directions  $P$ , the total number of parameters is actually less than the unfactored model.

### C. Filter Analysis

To better understand what the  $\text{tConv1}$  layer learns, Fig. 8 plots two-channel filter coefficients and the corresponding spatial responses, or beampatterns, after training.

Despite the intuition described in Section IV, the first layer filters appear to perform both spatial and spectral filtering. However, the beampatterns can nevertheless be categorized into a few broad classes. For example, filters 2, 3, 5, 7, and 9 in Fig. 8 only pass through some low frequency subbands below about 1.5 kHz, where most vowel energy occurs, but steered to have nulls in different directions. Very little spatial filtering is done in high-frequency regions, where many fricatives and stops occur. The low frequencies are most useful for localization because they are not subject to spatial aliasing and because they contain much of the energy in the speech signal; perhaps that is why the network exhibits this structure.

TABLE VI  
WER FOR TRAINING VS. FIXING THE  $\text{tConv1}$  LAYER, 2 CHANNEL

# Spatial Filters $P$	$\text{tConv1}$ Layer	WER
5	fixed	21.9
5	trained	20.7

TABLE VII  
FACTORED MODEL WER AFTER SEQUENCE TRAINING, SIMULATED

Method	WER - CE	WER - Seq
raw, unfactored, 2ch	21.8	18.2
raw, factored, 2ch	20.4	17.2
raw, unfactored, 4ch	20.8	17.2
raw, factored, 4ch	19.6	16.3

To further understand the benefit of the spatial and spectral filtering in  $\text{tConv1}$ , we enforce this layer to only perform spatial filtering by initializing the filters to be an impulse centered at a delay of zero for channel 0, and offset from zero in channel 1 by different delays for each filter. By not training this layer, this amounts to performing delay-and-sum filtering across a set of fixed look directions. Table VI compares performance when fixing vs. training the  $\text{tConv1}$  layer. The results demonstrate that learning the filter parameters, and therefore performing some spectral decomposition, improves performance over keeping this layer fixed.

### D. Results Summary

To conclude this section, Table VII shows the results after sequence training, comparing the factored and unfactored models. Notice that the 2 channel factored model provides 6% relative improvement over the unfactored model, while the 4 channel model provides 5% relative improvement. We do not go above 4 channels, as results from Table IV in Section III-F show that there is no difference between 4 and 8 channels.

## V. ADAPTIVE BEAMFORMING

While the unfactored model improves over the factored model, the model also suffers from a few drawbacks. First, the learned filters in this model are fixed during decoding, which potentially limits the ability of these models to adapt to previously unseen or changing conditions. In addition, since the factored model must perform spectral filtering for every look direction, this comes with a large computational complexity.

### A. NAB Model

To address the limited adaptability and reduce the computational complexity of the models from [16], [26], we propose a neural network adaptive beamforming (NAB) model [30] which re-estimates a set of spatial filter coefficients at each input frame using a neural network.

The NAB model is depicted in Fig. 9. At each time frame  $l$ , it takes in a small window of  $M$  waveform samples



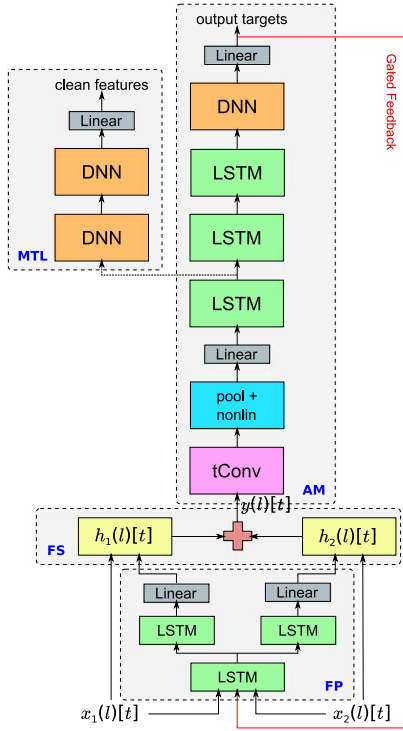


Fig. 9. Neural network adaptive beamforming (NAB) model architecture. It consists of filter prediction (FP), filter-and-sum (FS) beamforming, acoustic modeling (AM) and multi-task learning (MTL). The figure shows only two channels just for simplicity.

for each channel  $c$  from the  $C$  channel inputs, denoted as  $x_0(l)[t], x_1(l)[t], \dots, x_{C-1}(l)[t]$  for  $t \in \{0, \dots, M-1\}$ . Additionally to previous notations, the frame index  $l$  is explicitly used in this section to emphasize the frame dependent filtering coefficients. For simplicity, the figure shows an NAB model with  $C = 2$  channels. We will describe the different modules of this model in subsequent subsections.

Note that our NAB model is similar to the model proposed in [31], although filtering was performed in the frequency domain, as opposed to our model which processes time domain signals. We will show in Section VI-D that performing NAB in the time domain requires estimation of many fewer filter coefficients, and results in better WER compared to frequency domain filter prediction.

1) *Adaptive Filters*: The adaptive filtering layer is given by (4), where  $h_c(l)[n]$  is the estimated filter for channel  $c$  at time frame  $l$ . This model is very similar to the FS model from (1), except now the steering delay  $\tau_c$  is implicitly absorbed into the estimated filter parameters.

$$y(l)[t] = \sum_{c=0}^{C-1} \sum_{n=0}^{N-1} h_c(l)[n] x_c(l)[t-n] \quad (4)$$

In order to estimate  $h_c(l)[t]$ , we train a filter prediction (FP) module with one shared LSTM layer, one layer of channel-dependent LSTMs and linear output projection layers to predict  $N$  filter coefficients for each channel. The input to the *FP module* is a concatenation of frames of raw input samples  $x_c(l)[t]$  from all the channels, and can also include features typically

computed for localization such as cross correlation features [31]–[33]. The estimation of *FP module* parameters are jointly done with AM parameters by directly minimizing a cross-entropy or sequence loss function. Following (4) the estimated filter coefficients  $h_c(l)[t]$  are convolved with input samples  $x_c(l)[t]$  for each channel. The outputs of the convolution are summed across channels to produce a single channel signal  $y(l)[t]$ .

After adaptive FS, the single channel enhanced signal  $y(l)[t]$  is passed to an *AM module* (Fig. 9). We adopt the single channel raw waveform CLDNN model [15] for acoustic modeling, except that we now skip the frequency convolution layer as it has recently been shown in [34] to not help for noisier tasks. During training, the AM and FP (Fig. 9) are trained jointly.

2) *Gated Feedback*: Augmenting the network input at each frame with the prediction from the previous frame has been shown to improve performance [35]. To investigate the benefit of feedback in the NAB model, we pass the AM prediction at frame  $l-1$  back to the FP model at time frame  $l$  (red line in Fig. 9). Since the softmax prediction is very high dimensional, we feed back the low-rank activations preceding the softmax to the *FP module* to limit the increase of model parameters [36].

This feedback connection gives the *FP module* high level information about the phonemic content of the signal to aid in estimating beamforming filter coefficients. This feedback is comprised of model *predictions* which may contain errors, particularly early in training, and therefore might lead to poor model training [35]. A gating mechanism [37] is hence introduced to the connection to modulate the degree of feedback. Unlike conventional LSTM gates, which control each dimension independently, we use a global scalar gate to moderate the feedback. The gate  $g^{\text{fb}}(l)$  at time frame  $l$ , is computed from the input waveform samples  $x(l)$ , the state of the first FP LSTM layer  $s(l-1)$ , and the feedback vector  $v(l-1)$ , as follows:

$$g^{\text{fb}}(l) = \sigma(w_x^T \cdot x(l) + w_s^T \cdot s(l-1) + w_v^T \cdot v(l-1)) \quad (5)$$

where  $w_x$ ,  $w_s$  and  $w_v$  are the corresponding weight vectors and  $\sigma$  is an elementwise non-linearity. We use a logistic function for  $\sigma$  which outputs values in the range  $[0, 1]$ , where 0 cuts off the feedback connection and 1 directly passes the feedback through. The effective FP input is hence  $[x(l), g^{\text{fb}}(l)v(l-1)]$ .

3) *Regularization with MTL*: Multi-task learning has been shown to yield improved robustness [26], [38], [39]. We adopt an *MTL module* similar to [26] during training by configuring the network to have two outputs, one recognition output which predicts CD states and a second denoising output which reconstructs 128 log-mel features derived from the underlying clean signal. The denoising output is only used in training to regularize the model parameters; the associated layers are discarded during inference. In the NAB model the *MTL module* branches off of the first LSTM layer of the *AM module*, as shown in Fig. 9. The *MTL module* is composed of two fully connected DNN layers followed by a linear output layer which predicts clean features. During training the gradients back propagated from the two outputs are weighted by  $\alpha$  and  $1 - \alpha$  for the recognition and denoising outputs respectively.

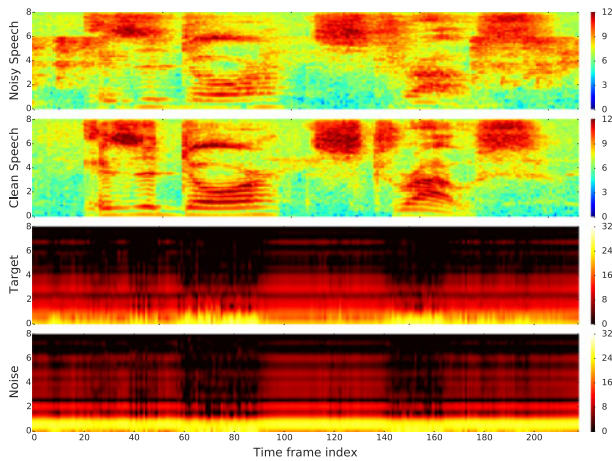


Fig. 10. Visualizations of the predicted beamformer responses at different frequency (Y-axis) across time (X-axis) at the target speech direction (3rd) and interfering noise direction (4th) with the noisy (1st) and clean (2nd) speech spectrograms.

### B. NAB Filter Analysis

The best NAB model found in [30] has the following configurations:

- 1) the *FP module* has one shared 512-cell LSTM layer across channels, one layer of channel-dependent 256-cell LSTMs and one layer of channel-dependent 25-dimensional linear projection layer;
- 2) the *FP module* takes in the concatenation of raw waveform samples from each channel;
- 3) the *FP module* outputs a 1.5 ms filter (25-dimensional vector) for each channel;
- 4) the *AM module* is a single channel raw waveform LDNN model [15] with 256 tConv filters and without the frequency convolution layer [34], which is also similar to other multichannel models discussed in this paper;
- 5) 128-dimensional clean log-mel features are used as the secondary reconstruction objectives with a weight of 0.1 for MTL;
- 6) per-frame gated feedback connection from the bottleneck layer right before the *AM module*'s softmax layer is appended to the *FP module*'s input.

Fig. 10 illustrates the frequency responses of the predicted beamforming filters at the target speech and interfering noise directions. The SNR for this utterance is 12 dB. The responses in the target speech direction have relatively more speech-dependent variations than those in the noise direction. This may indicate that the predicted filters are attending to the speech signal. Besides, the responses at high speech-energy regions are generally lower than others, which suggests the automatic gain control effect of the predicted filters.

### C. Result Summary

Finally, to conclude this section, we show the results after sequence training compared to the factored model. To understand the impact of WER with respect to each factor discussed

TABLE VIII  
COMPARISON BETWEEN 2-CHANNEL FACTORED AND ADAPTIVE MODELS, IN TERMS OF WER AT CROSS-ENTROPY (CE) AND SEQUENCE (SEQ) TRAINING, AS WELL AS THE TOTAL NUMBER OF MULTIPLIES AND ADDS (MULTADD) IN MILLIONS

Model	WER (%)		Param	MultAdd
	CE	Seq		
factored (w/MTL)	20.1	16.9	18.9M	35.1M
NAB (w/MTL)	20.5	17.2	24.0M	28.8M

in Section V, we refer the reader to [30]. Instead, in this paper we show the results for the best NAB setup, which uses the factors outlined in Section V-B. Since the NAB model is trained without frequency convolution (i.e., LDNN), we do the same for the factored model. In addition, we show results for the factored model with MTL to be comparable to the NAB model. Table VIII shows that the factored model can potentially handle different directions by enumerating many look directions in the spatial filtering layer and can achieve slightly better performance compared to the adaptive model. However, the adaptive model has less computational complexity, as measured by the number of multiplies and additions (MultAdd) of the model, as shown in the Table.

## VI. FILTERING IN THE FREQUENCY DOMAIN

Until now, we have presented three multichannel models in the time domain. However, it is well known that “circular” convolution between two time domain signals is equivalent to the element-wise product of their frequency domain counterparts [40], [41]. A benefit of operating in the complex FFT space is that element-wise products are much faster to compute compared to convolutions, particularly when the convolution filters and input size is large as in our multichannel raw waveform models. In this section, we describe how we can implement both the factored Model from Section IV and the NAB Model from Section V, in the frequency domain [29].

### A. Factored Model

In this section, we describe the factored model in the frequency domain.

1) *Spatial Filtering*: For frame index  $l$  and channel  $c$ , we denote  $X_c[l] \in \mathbb{C}^K$  as the result of an  $M$ -point Fast Fourier Transform (FFT) of  $x_c[t]$  and  $H_c^p \in \mathbb{C}^K$  as the FFT of  $h_c^p$ . Note that we ignore negative frequencies because the time domain inputs are real, and thus our frequency domain representation of an  $M$ -point FFT contains only  $K = M/2 + 1$  unique complex-valued frequency bands. The spatial convolution layer in (2) can be represented by (6) in the frequency domain, where  $\cdot$  denotes element-wise product. We denote the output of this layer as  $Y^p[l] \in \mathbb{C}^K$  for each look direction  $p$ :

$$Y^p[l] = \sum_{c=0}^{C-1} X_c[l] \cdot H_c^p \quad (6)$$

In this paper, we explore two different methods for implementing the “spectral filtering” layer in the frequency domain.

2) *Spectral Filtering: Complex Linear Projection*: It is straightforward to rewrite the convolution in (3) as an element-wise product in frequency, for each filter  $f$  and look direction  $p$ :

$$W_f^p[l] = Y^p[l] \cdot G_f \quad (7)$$

In the above equation,  $W_f^p[l] \in \mathbb{C}^K$  and  $G_f \in \mathbb{C}^K$  is the FFT of the time domain filter  $g_f$  in (3). There is no frequency domain equivalent to the max-pooling operation in the time domain. Therefore to mimic max-pooling exactly requires taking the inverse FFT of  $W_f^p[l]$  and performing the pooling operation in the time domain, which is computationally expensive to do for each look direction  $p$  and filter output  $f$ .

As an alternative [42] recently proposed the Complex Linear Projection (CLP) model which performs average pooling in the frequency domain and results in similar performance to a single channel raw waveform model. Similar to the waveform model the pooling operation is followed by a point-wise absolute-value non-linearity and log compression. The 1-dimensional output for look direction  $p$  and filter  $f$  is given by:

$$Z_f^p[l] = \log \left| \sum_{k=1}^N W_f^p[l, k] \right| \quad (8)$$

Collecting the output for all look directions and filters, the feature at frame index  $l$  is denoted as  $Z[l] \in \{Z_1^1[l], \dots, Z_F^P[l]\} \in \mathbb{R}^{1 \times F \times P}$ , with the same dimensions as the corresponding feature frame in the waveform model,  $z[l]$ .

3) *Spectral Filtering: Linear Projection of Energy*: We also explore an alternative decomposition that is motivated by the log-mel filterbank. Given the complex-valued FFT for each look direction,  $Y^p[l]$ , we first compute the energy at each time-frequency bin  $(l, k)$ :

$$\hat{Y}^p[l, k] = |Y^p[l, k]|^2 \quad (9)$$

After applying a power compression with  $\alpha = 0.1$ ,  $\hat{Y}^p[l]$  is linearly projected down to an  $F$  dimensional space, in a process similar to the mel filterbank, albeit with learned filter shapes:

$$Z_f^p[l] = G_f \times (\hat{Y}^p[l])^\alpha \quad (10)$$

As in the other models, the projection weights  $G \in \mathbb{R}^{K \times F}$ , are shared across all look directions.

The main difference between the CLP and LPE models is that the former retains phase information when performing the filterbank decomposition with matrix  $G$ . In contrast, LPE operates directly on the energy in each frequency band with the assumption that phase not important for computing features.

## B. NAB Model

In the frequency-domain NAB setup, we have an LSTM which predicts complex FFT (CFFT) inputs for both channels. Given a 512-pt FFT input, this amounts to predicting  $4 \times 257$  frequency points for real and imaginary components for 2 channels, which is much more than the predicted filter size in the time

TABLE IX  
FREQUENCY DOMAIN FACTORED MODEL PERFORMANCE, IN TERMS OF WER AT CROSS-ENTROPY (CE) AND SEQUENCE (SEQ) TRAINING, AS WELL AS THE TOTAL NUMBER OF MULTIPLIES AND ADDS (MULTADD) IN MILLIONS

Model	Spatial MultAdd	Spectral MultAdd	Total MultAdd	WER CE	WER Seq
time	525.6K	15.71M	35.1M	20.4	17.1
CLP	10.3K	655.4K	19.6M	20.5	17.2
LPE	10.3K	165.1K	19.1M	20.7	17.2

TABLE X  
RESULTS WITH A 64MS WINDOW SIZE, IN TERMS OF WER AT CROSS-ENTROPY (CE) AND SEQUENCE (SEQ) TRAINING, AS WELL AS THE TOTAL NUMBER OF MULTIPLIES AND ADDS (MULTADD) IN MILLIONS

Feat	Spatial MultAdd	Spectral MultAdd	Total MultAdd	WER Seq
time	906.1K	33.81M	53.6M	17.1
CLP	20.5K	1.3M	20.2M	17.1
LPE	20.5K	329.0K	19.3M	16.9

domain (i.e., 1.5 ms = 25 taps). After the complex filters are predicted for each channel, element-wise product is done with the FFT of the input for each channel, mimicking the convolution in (4) in the frequency domain. The output of this is given to a single channel LDNN in the frequency domain, which does spectral decomposition, using either CLP or LPE, and acoustic modeling.

## C. Results: Factored Model

1) *Performance*: First, we explore the performance of the frequency domain factored model. Note this model does not have any frequency convolution layer. We explore this for a similar setting to most efficient raw-waveform factored setup [29], namely  $P = 5$  look directions in the spatial layer and  $F = 128$  filters in the spectral layer. The input is 32 ms instead of 35 ms like raw-waveform, as this allows us to take a  $D = 512$ -point DFT without zero-padding at a sampling rate of 16 kHz. A 35-ms input would have required us to take a 1024-point DFT, and we have not found any big difference in performance between 32 and 35 ms inputs for raw-waveform.

Table IX shows that the WER performance of both the CLP and LPE factored models are similar. The table also indicates the total number of multiplication and addition operations (MultAdd) for different layers of the model. Both models reduce the number of operations by a factor of 1.9x over the best waveform model, with a small degradation in WER.

However, given that the frequency models are more computationally efficient, we explore improving WER by increasing the window size (and therefore computational complexity) of the factored models. Specifically, since longer windows typically help with localization [6], we explore using 64ms input windows for both models. With a 64ms input, the frequency models require a 1024-point FFT. Table X shows that the frequency models improve the WER over using a smaller 32 ms input, and



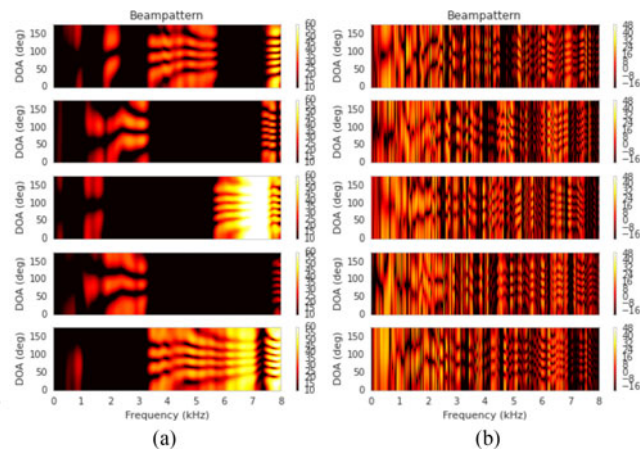


Fig. 11. Beampatterns of time and frequency models. (a) Factored model, time, (b) Factored model, frequency.

still perform roughly the same. However, the frequency model now has an even larger computational complexity savings of 2.7x savings compared to the time domain model.

2) *Comparison Between Learning in Time vs. Frequency:* Fig. 11 shows the spatial responses (i.e., beampatterns) for both the time and frequency domain spatial layers. Since the LPE and CLP models have the same spatial layer and we have found the beampatterns to look similar, we only plot the CLP model for simplicity. The beampatterns show the magnitude response in dB as a function of frequency and direction of arrival, i.e. each horizontal slice of the beampattern corresponds to the filter’s magnitude response for a signal coming from a particular direction. In each frequency band (vertical slice), lighter shades indicate that sounds from those directions are passed through, while darker shades indicate directions whose energy is attenuated. The figures show that the spatial filters learned in the time domain are band-limited, unlike those learned in the frequency domain. Furthermore, the peaks and nulls are aligned well across frequencies for the time domain filters. One hypothesis we have for the band-limited nature of the raw waveform spatial filters is that they are short (i.e., 5 ms), compared to the frequency-domain spatial filters which span over the entire 32 or 64-ms input.

The differences between these models can further be seen in the magnitude responses of the spectral layer filters, as well as in the outputs of the spectral layers from different look directions plotted for an example signal. Fig. 12 illustrates that the magnitude responses in both time and CLP models look qualitatively similar, and learn bandpass filters with increasing center frequency. However, because the spatial layers in time and frequency are quite different, we see that the spectral layer outputs in time are much more diverse in different spatial directions compared to the CLP model. In contrast to these models, the LPE spectral layer does not seem to learn bandpass filters.

At some level, time-domain and frequency-domain representations are interchangeable, but they result in networks that are parameterized very differently. Even though the time and frequency models all learn different spatial filters, they all seem to have similar WERs. In addition, even though the spatial layer

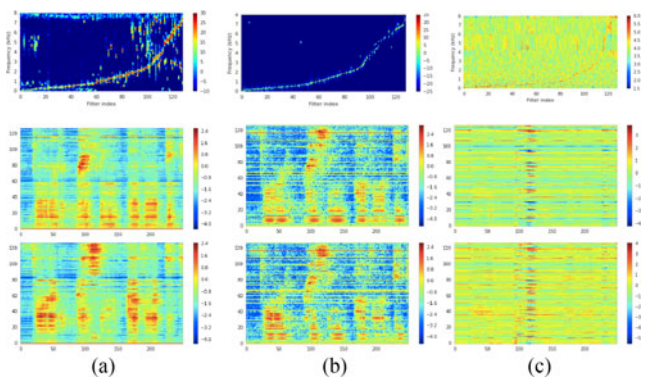


Fig. 12. The first row shows the magnitude response of the spectral layer for the (a) raw-waveform, (b) CLP and (c) LPE models. The second and third rows show the output of the spectral layers of these models for two different look directions.

TABLE XI  
COMPARISON BETWEEN TIME AND FREQUENCY NAB MODELS

Model	WER (%)	Param (M)	MultAdd (M)
raw	20.4	18.9	35.1
CLP	21.0	24.7	25.1

of the CLP and LPE models are different, they too seem to have similar performance. There are roughly 18M parameters in the LDNN model that sits above the spatial/spectral layers, which accounts for over 90% of the parameters in the model. Any differences between the spatial layers in time and frequency are likely accounted for in the LDNN part of the network.

#### D. Results: Adaptive Model

Next, we explore the performance of the frequency domain NAB model. Table XI shows the WER at CE and computational complexity of the raw-waveform and CLP NAB models. While using CLP features greatly reduces computational complexity, the performance is worse than the raw-waveform model. One hypothesis we have is that frequency domain processing requires predicting a higher dimensional filter, which we can see from the table leads to a degradation in performance. Since the CLP and LPE models perform similarly for the factored model, and the CLP NAB model degraded performance over raw NAB, we did not repeat the experiment for LPE NAB.

### VII. FINAL COMPARISON, RE-RECORDED DATA

Finally, to show that a model trained on simulated data can handle real reverberation, we also evaluated the performance of different multichannel models presented in this paper on a real “Rerecorded” test set. Reverberation-I is when the microphone is placed on a coffee table, whereas Reverberation-II is when the mic is placed on a TV stand. Since this set contains a circular microphone geometry but our models are trained on a linear microphone geometry, we only report results with 2 channels to form a linear array with a 7.5 cm spacing (in contrast to the 14 cm spacing during training). In spite of these mismatches in



TABLE XII  
WER ON “RERECORDED” SET AFTER SEQUENCE TRAINING

Model	Rev.-I	Rev.-II	Rev.-I Noisy	Rev.-II Noisy	Ave
1 channel raw	18.6	18.5	27.8	26.7	22.9
2 channel raw, unfactored	17.9	17.6	25.9	24.7	21.5
2 channel raw, factored	17.1	16.9	24.6	24.2	20.7
2 channel LPE, factored	17.4	16.8	25.2	23.5	20.7
2 channel raw, NAB	17.8	18.1	27.1	26.1	22.3
7 channel, superdirective	-	-	25.3	23.7	-

reverberation and in microphone spacing, our technique performs well.

Table XII shows the results with different multichannel models after sequence training. All raw-waveform models are trained with 35-ms inputs and 128 spectral decomposition filters. The factored model has 5 look directions. The LPE factored model is trained with a 64-ms input, 5 look directions, and 128 spectral decomposition filters. All frontends use an LDNN architecture in the upper layers of the network.

Notice that the 2 channel raw factored model gives a 13% relative improvement over single channel, with larger improvements in noisier test sets, which is to be expected. In addition, the LPE factored model performs similar to the raw factored model. Finally, the NAB model performs much worse than the factored model. We have not investigated this performance difference in detail, but one hypothesis is that because the NAB model adapts its filters over the course of an utterance, it may sometimes adapt its filters into a bad configuration from which it cannot recover when presented with mismatched test data.

We also include results on the noisier reverberation sets with oracle localization and a robust superdirective beamformer similar to [43], [44] with oracle knowledge of the speech and noise. Note that our superdirective beamformer is a fixed filter and sum beamformer similar in spirit to a maximum signal-to-noise ratio filter [2]. The table shows that for the noisier reverberation sets, our factored models, with just 2 channels and no oracle information, are able to match the performance of the oracle superdirective beamformer with 7 channels.

It is important to note that the motivation of the above results is to show the robustness of the neural network methods when there is slight mismatch in microphone array geometry and noise conditions between training and test, as this condition matches the real-world scenarios we currently deal with. For very severe microphone array geometry mismatch, we have noticed a larger degradation in performance, as shown in Section 4.4 of our paper [16]. In order to address severe mismatch, we have found that we must train the system with a variety of microphone spacings to make the model more robust.

## VIII. CONCLUSIONS AND FUTURE WORK

In this paper, we introduced a methodology to do multichannel enhancement and acoustic modeling jointly within a neural network framework. First, we developed an unfactored raw-waveform multichannel model, and showed that this model

performed as well as a model given oracle knowledge of the true location. Next, we introduced a factored multichannel model to separate out spatial and spectral filtering operations, and found that this offered an improvement over the unfactored model. Next, we introduced an adaptive beamforming method, which we found to match the performance of the multichannel model with far fewer computations. Finally, we showed that we can match the performance of the raw-waveform factored model, with far fewer computations, with a frequency-domain factored model. Overall, the factored model provides between a 5–13% relative improvement over single channel and traditional signal processing techniques, on both simulated and rerecorded test sets.

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