MEGATRON-LM: TRAINING MULTI-BILLION PARAMETER LANGUAGE MODELS USING MODEL PARALLELISM

Mohammad Shoeybi [1 2] **Mostofa Patwary** [1 2] **Raul Puri** [1 2] **Patrick LeGresley** [2] **Jared Casper** [2]
Bryan Catanzaro [2]

ABSTRACT

Recent work in unsupervised language modeling demonstrates that training large neural language models advances the state of the art in Natural Language Processing applications. However, for very large models, memory constraints limit the size of models that can be practically trained. Model parallelism allows us to train larger models, because the parameters can be split across multiple processors. In this work, we implement a simple, efficient intra-layer model parallel approach that enables training state of the art transformer language models with billions of parameters. Our approach does not require a new compiler or library changes, is orthogonal and complimentary to pipeline model parallelism, and can be fully implemented with the insertion of a few communication operations in native PyTorch. We illustrate this approach by converging an 8.3 billion parameter transformer language model using 512 GPUs, making it the largest transformer model ever trained at 24x times the size of BERT and 5.6x times the size of GPT-2. We sustain up to 15.1 PetaFLOPs per second across the entire application with 76% scaling efficiency, compared to a strong single processor baseline that sustains 39 TeraFLOPs per second, which is 30% of peak FLOPs. The model is trained on 174GB of text, requiring 12 ZettaFLOPs over 9.2 days to converge. Transferring this language model achieves state of the art (SOTA) results on the WikiText103 (10.8 compared to SOTA perplexity of 16.4) and LAMBADA (66.5% compared to SOTA accuracy of 63.2%) datasets. We release training and evaluation code, as well as the weights of our smaller portable model, for reproducibility.

1 INTRODUCTION

Natural Language Processing (NLP) is advancing quickly, in part due to an increase in available compute and dataset size. The abundance of compute and data enables training increasingly larger language models via unsupervised language model pretraining (Devlin et al., 2018; Radford et al., 2019b). Empirical evidence indicates that larger language models are dramatically more useful for NLP tasks such as article completion, question answering, and natural language inference. By transferring or finetuning these pretrained language models on downstream natural language tasks, one can achieve state of the art results as shown in recent work (Devlin et al., 2018; Peters et al., 2018; Howard & Ruder, 2018; Radford et al., 2018; 2019b; 2017; Ramachandran et al., 2016; Liu et al., 2019b; Dai et al., 2019; Yang et al., 2019; Liu et al., 2019a).

As these models become larger, they exceed the memory limit of modern processors, and require additional memory management techniques such as activation checkpointing

---
[1]Equal contribution [2]NVIDIA Corporation. Correspondence to: {mshoeybi, mpatwary, raulp, plegresley, jcasper, bcatanzaro}@nvidia.com .

(Chen et al., 2016). Widely used optimization algorithms such as ADAM require additional memory per parameter to store momentum and other optimizer state, which reduces the size of models that can be effectively trained. Several approaches to model parallelism overcome this limit by partitioning the model such that the weights and their associated optimizer state do not need to reside concurrently on the processor. For example, GPipe (Huang et al., 2018) and Mesh-Tensorflow (Shazeer et al., 2018) provide frameworks for model parallelism of different kinds. However, they require rewriting the model, and rely on custom compilers and frameworks that are still under development and not applicable to all problems.

In this work, we train a transformer-based language model with 8.3 billion parameters efficiently using intra-layer model-parallelism. We exploit the inherent structure in transformer based language models to make a simple model-parallel implementation that trains efficiently in PyTorch, with no custom C++ code or compiler required. This approach is orthogonal to pipeline-based model parallelism as advocated by approaches such as (Huang et al., 2018).

To demonstrate the scalability of our approach, we establish a baseline by training a model of 1.2 billion parameters
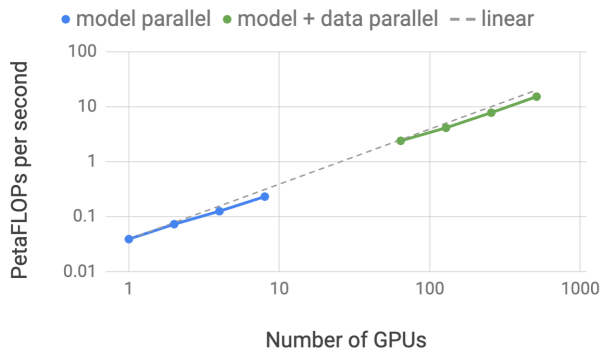
Figure 1. Model (blue) and model+data (green) parallel FLOPS as a function of number of GPUs. Model parallel (blue): up to 8-way model parallel weak scaling with approximately 1 billion parameters per GPU (e.g. 2 billion for 2 GPUs and 4 billion for 4 GPUs). Model+data parallel (green): similar configuration as model parallel combined with 64-way data parallel.

on a single NVIDIA V100 32GB GPU, that sustains 39 TeraFLOPs per second over the course of the entire training application. This is 30% of the theoretical peak FLOPS for a single GPU as configured in a DGX-2H server, and thus represents a very strong baseline. Scaling the model to 8.3 billion parameters on 512 GPUs with 8-way model parallelism, we achieve up to 15.1 PetaFLOPs per second sustained over the entire application. This is 76% scaling efficiency compared to the single GPU case. Converging the model on 174 GB of text over 9.2 days requires 12 ZettaFLOPs in total. Figure 1 shows more detailed scaling results.

We analyze the accuracy of our trained models by computing perplexity on the WikiText103 dataset and cloze-style prediction accuracy on the LAMBADA dataset. We show that the WikiText103 perplexity decreases and LAMBADA accuracy increases with increasing model size and achieves state of the art results on these tasks.

In summary, our contributions are as follows:

- We implement a simple and efficient model parallel approach by making only a few targeted modifications to an existing PyTorch transformer implementation.

- We demonstrate convergence of an 8.3 billion parameter Transformer-based Language Model, which is the largest transformer-based neural language model that has been published.

- We showcase that our models further advance the state of the art (SOTA) in natural language processing by achieving SOTA perplexity on WikiText103 (10.8 ppl) and SOTA accuracy on the LAMBADA dataset (66.5%).

- We perform an in-depth empirical analysis of our model and data parallel technique and demonstrate up to 76% scaling efficiency using 512 GPUs.

- We open source our code along with the training and evaluation pipelines at `https://github.com/nvidia/megatron-lm`

2 BACKGROUND AND CHALLENGES

2.1 Neural Language Model Pretraining

Pretrained language models have become an indispensable part of NLP researchers' toolkits. Leveraging large corpus pretraining to learn robust neural representations of language is an active area of research that has spanned the past decade. Throughout this period, the community has seen an increasing trend in the scale and complexity of these pretraining methods that have steadily advanced the state of the art. Early examples of pretraining and transferring neural representations of language demonstrated that pretrained word embedding tables improve downstream task results compared to word embedding tables learned from scratch (Mikolov et al., 2013; Pennington et al., 2014; Turian et al., 2010). Later work advanced research in this area by learning and transferring neural models that capture contextual representations of words (Melamud et al., 2016; McCann et al., 2017; Peters et al., 2018; Radford et al., 2017; 2019b). Recent parallel work (Ramachandran et al., 2016; Howard & Ruder, 2018; Radford et al., 2018; Devlin et al., 2018; Liu et al., 2019b; Dai et al., 2019; Yang et al., 2019; Liu et al., 2019a) further builds upon these ideas by not just transferring the language model to extract contextual word representations, but by also finetuning the language model in an end to end fashion on downstream tasks.

Through these works, the state of the art has advanced from transferring just word embedding tables to transferring entire 1.5B parameter language models. This progression of methods has necessitated the need for hardware, systems techniques, and frameworks that are able to operate efficiently at scale and satisfy increasing computational needs. Our work aims to provide the tools necessary to take another step forward in this trend.

2.2 Transformer Language Models and Multi-Head Attention

Language modeling is a central task in natural language processing and language understanding. It is widely used in many applications such as speech recognition, question and answering, and summarization. To model the sequential nature of language, recurrent neural networks (RNNs/LSTMs (Hochreiter & Schmidhuber, 1997)) have been used for more than a decade. However, due to their inability to model long range dependencies and the sequential nature of
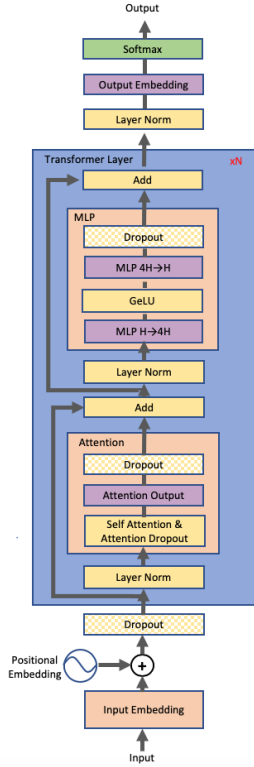
Figure 2. GPT-2 Transformer Architecture. Purple blocks correspond to fully connected layers. Each blue block represents a single transformer layer that is replicated N times.

these models (processes tokens one by one), efficient training on large corpora has been challenging. Recently, new approaches based on attention modules, called *transformers* (Vaswani et al., 2017), have been introduced. These models demonstrate superior accuracy and compute efficiency. Instead of considering tokens one by one and maintaining a hidden state, transformers consider entire segments of tokens and learn how to interpret the dependencies within them.

The original *transformer* formulation was designed as a machine translation architecture that transforms an input sequence into another output sequence using two parts, an *Encoder* and *Decoder*. However, recent works leveraging transformers for language modeling such as GPT (Radford et al., 2018), BERT (Devlin et al., 2018) and GPT-2 (Radford et al., 2019b) use only the *Encoder* or *Decoder* depending on their needs. GPT, GPT-2, and other autoregressive transformer language models (Dai et al., 2019; Yang et al., 2019) employ multi-layer *transformer decoder* architectures. Our work focuses on architectures similar to GPT-2. A high level overview of the GPT-2 *transformer* model is presented below.

Figure 2 shows a schematic diagram of the model we used. The model consists of an input subword token embedding layer, positional embedding layer, stack of identical trans-

former layers, and linear output embedding layer followed by a final softmax layer. The embedding layer embeds the input subword tokens into vectors and positional encoding helps the transformer capture the order of the input tokens. The positional encoding is added to the output of the subword embedding layer, and fed to the first transformer layer. The normalized output of the last transformer layer is given to a linear layer with weights equivalent to the transpose of the input embedding. To generate probability distributions for output subword tokens a final softmax layer is applied.

As mentioned above, each transformer layer is a decoder only transformer block, which consist of a multi-head attention layer with a left-to-right attention mask, and a feed forward layer. Each multi-head attention layer consists of several attention heads that run in parallel. Each attention head uses self attention to adaptively process each token input conditioned on the other input tokens. The left-to-right attention mask ensures that a given input only attends to the positions that precede it to the left. Each head uses a fully connected layer to map each token to a key, query, and value vector of dimension *hidden-size/number-of-attention-heads*. Each attention head then maps a query over all key-value pairs to its output. For a single query in the sequence, this is implemented as a scaled dot product of the query with all other keys followed by a softmax that is then used to obtain a weighted sum over all values. To improve robustness of the model and aid in training convergence, attention dropout is applied to the softmax values. For a single token, the outputs of the multiple attention heads are concatenated into a vector that is the size of the hidden dimension. More details on the transformer layer can be found in (Vaswani et al., 2017).

It is worthwhile to mention that GPT-2 uses GeLU (Hendrycks & Gimpel, 2016) nonlinearities and layer normalization (Ba et al., 2016) to the input of the multi-head attention and feed forward layers, whereas the original transformer (Vaswani et al., 2017) uses ReLU nonlinearities and applies layer normalization to outputs. Therefore, in the GPT-2 model, an additional layer normalization is added after the final transformer layer.

2.3 Data and Model Parallelism in Deep Learning

There are two central paradigms for scaling out deep neural network training to numerous hardware accelerators: data parallelism (Valiant, 1990) where a training minibatch is split across multiple workers, and model parallelism in which the memory usage and computation of a model is distributed across multiple workers. Data parallelism has become an indispensable tool for large scale deep neural network training due to its desirable weak scaling properties. By increasing the minibatch size proportionally to the number of available workers, one observes near linear scaling

in training data throughput. However, large batch training introduces complications into the optimization process that can result in reduced accuracy or longer time to convergence, offsetting the benefit of increased training throughput (Keskar et al., 2017). Further research (Goyal et al., 2017; You et al., 2017; 2019) has developed techniques to mitigate these effects and drive down training time of large neural networks from weeks and months to the order of days, hours, minutes and in some cases even seconds. To scale out training even further, parallel work (Chen et al., 2016) has combined data parallelism with activation checkpointing: recomputing activations in the backward pass without storing them in the forward pass to reduce memory requirements.

However, these techniques have one fundamental limitation in the problem size they can tackle: the model must fit entirely on one worker. With language models of increasing size and complexity like BERT and GPT-2, neural networks have approached the memory capacity of modern hardware accelerators. To continue advancing the field and train larger language models that scale well with computational resources, we must utilize model parallelism in addition to data parallelism. Within model parallelism, there are two further paradigms: layer-wise pipeline parallelism, and more general distributed tensor computation. In pipeline model parallelism, groups of operations are performed on one device before the outputs are passed to the next device in the pipeline where a different group of operations are performed. To ensure that the devices are not idle, waiting for input from other devices, pipeline parallel approaches partition (and usually increase) the batch size so that a portion of the minibatch is always being computed on a device, and all devices are being maximally utilized at all times. This approach to model parallelism mirrors the instruction pipelining found in CPUs. Some approaches (Harlap et al., 2018; Chen et al., 2018) use a parameter server (Li et al., 2014) in conjunction with pipeline parallelism. However these suffer from inconsistency issues. The GPipe framework for TensorFlow (Huang et al., 2018) overcomes this inconsistency issue by using synchronous gradient decent. This approach requires additional logic to handle the efficient pipelining of these communication and computation operations, and suffers from pipeline bubbles that reduce efficiency, or changes to the optimizer itself which impact accuracy.

Distributed tensor computation is a more general approach that partitions a tensor operation across multiple devices to accelerate computation or increase model size. FlexFlow (Jia et al., 2018), a deep learning framework orchestrating such parallel computation, provides a method to pick the best parallelization strategy. Recently, Mesh-TensorFlow (Shazeer et al., 2018) introduced a language for specifying a general class of distributed tensor computations in TensorFlow (Abadi et al., 2015). The parallel dimensions are specified in the language by the end user and the resulting graph is compiled with proper collective primitives. We utilize similar insights to those leveraged in Mesh-TensorFlow and exploit parallelism in computing the transformer's attention heads to parallelize our transformer model. However, rather than implement a framework and compiler for model parallelism, we make only a few targeted modifications to existing PyTorch transformer implementations. Our approach is simple, does not require any new compiler or code re-wiring, and can be fully implemented by inserting a few simple primitives, as described in the next section.

3 MODEL PARALLEL TRANSFORMERS

We take advantage of the structure of transformer networks to create a simple model parallel implementation by adding a few synchronization primitives. A transformer layer consists of a self attention block followed by a two-layer, multi-layer perceptron (MLP) as shown in Figure 2. We introduce model parallelism in both of these blocks separately.

We start by detailing the MLP block. The first part of the block is a GEMM followed by a GeLU nonlinearity:

$$Y = \text{GeLU}(XA) \tag{1}$$

One option to parallelize the GEMM is to split the weight matrix A along its rows and input X along its columns as:

$$A = \begin{bmatrix} A_1 \\ A_2 \end{bmatrix}, \ X = [X_1, X_2]. \tag{2}$$

This partitioning will result in $Y = \text{GeLU}(X_1 A_1 + X_2 A_2)$. Since GeLU is a nonlinear function, $\text{GeLU}(X_1 A_1 + X_2 A_2) \neq \text{GeLU}(X_1 A_1) + \text{GeLU}(X_2 A_2)$ and this approach will require a synchronization point before the GeLU function.

Another option is to split A along its columns $A = [A_1, A_2]$. This partitioning allows the GeLU nonlinearity to be independently applied to the output of each partitioned GEMM:

$$[Y_1, Y_2] = [\text{GeLU}(XA_1), \text{GeLU}(XA_2)] \tag{3}$$

This is advantageous as it removes a synchronization point. Hence, we partition the first GEMM in this column parallel fashion and split the second GEMM along its rows so it takes the output of the GeLU layer directly without requiring any communication as shown in Figure 3a. The output of the second GEMM is then reduced across the GPUs before passing the output to the dropout layer. This approach splits both GEMMs in the MLP block across GPUs and requires only a single all-reduce operation in the forward pass (g operator) and a single all-reduce in the backward pass (f operator). These two operators are conjugates of each other and can be implemented in PyTorch with only a few lines of code. As an example, the implementation of the f operator is provided below:
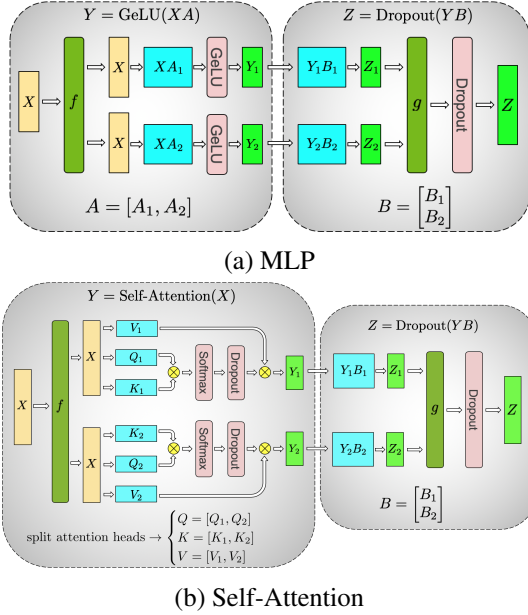
(a) MLP

(b) Self-Attention

Figure 3. Blocks of Transformer with Model Parallelism. f and g are conjugate. f is an identity operator in the forward pass and all reduce in the backward pass while g is an all reduce in the forward pass and identity in the backward pass.

```
class f(torch.autograd.Function):
    def forward(ctx, x):
        return x
    def backward(ctx, gradient):
        all_reduce(gradient)
        return gradient
```

Code 1. Implementation of f operator. g is similar to f with identity in the backward and all-reduce in the forward functions.

As shown in Figure 3b, for the self attention block we exploit inherent parallelism in the multihead attention operation, partitioning the GEMMs associated with key (K), query (Q), and value (V) in a column parallel fashion such that the matrix multiply corresponding to each attention head is done locally on one GPU. This allows us to split per attention head parameters and workload across the GPUs, and doesn't require any immediate communication to complete the self-attention. The subsequent GEMM from the output linear layer (after self attention) is parallelized along its rows and takes the output of the parallel attention layer directly, without requiring communication between the GPUs. This approach for both the MLP and self attention layer fuses groups of two GEMMs, eliminates a synchronization point in between, and results in better scaling. This enables us to perform all GEMMs in a simple transformer layer using only two all-reduces in the forward path and two in the backward path (see Figure 4).

The transformer language model has an output embedding with the dimension of hidden-size (H) times vocabulary-
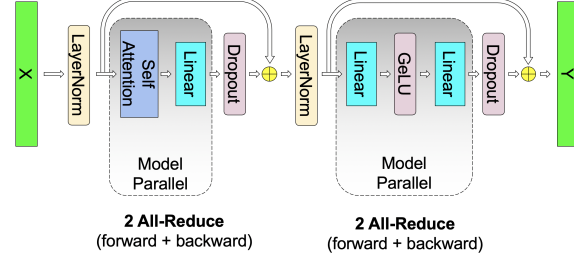

Figure 4. Communication operations in a transformer layer. There are 4 total communication operations in the forward and backward pass of a single model parallel transformer layer.

size (v). Since the vocabulary size is on the order of tens of thousands of tokens for modern language models (for example, GPT-2 used a vocabulary size of 50,257), it is beneficial to parallelize the output embedding GEMM. However, in transformer language models, the output embedding layer shares weights with the input embedding, requiring modifications to both. We parallelize the input embedding weight matrix $E_{H \times v}$ along the vocabulary dimension $E = [E_1, E_2]$ (column-wise). Since each partition now only contains a portion of the embedding table, an all-reduce (g operator) is required after the input embedding. For the output embedding, one approach is to perform the parallel GEMM $[Y_1, Y_2] = [XE_1, XE_2]$ to obtain the logits, add an all-gather $Y = \text{all-gather}([Y_1, Y_2])$, and send the results to the cross-entropy loss function. However, for this case, the all-gather will communicate $b \times s \times v$ elements (b is the batch-size and s is the sequence length) which is huge due to vocabulary size being large. To reduce the communication size, we fuse the output of the parallel GEMM $[Y_1, Y_2]$ with the cross entropy loss which reduces the dimension to $b \times s$. Communicating scalar losses instead of logits is a huge reduction in communication that improves the efficiency of our model parallel approach.

Much of our model parallel approach can be characterized as techniques aimed at reducing communication and keeping the GPUs compute bound. Rather than having one GPU compute part of the dropout, layer normalization, or residual connections and broadcast the results to other GPUs, we choose to duplicate the computation across GPUs. Specifically, we maintain duplicate copies of layer normalization parameters on each GPU, and take the output of the model parallel region and run dropout and residual connection on these tensors before feeding them as input to the next model parallel regions. To optimize the model we allow each model parallel worker to optimize its own set of parameters. Since all values are either local to or duplicated on a GPU, there is no need for communicating updated parameter values in this formulation.

In summary, our approach as described above is simple to implement, requiring only a few extra all-reduce operations

added to the forward and backward pass. It does not require a compiler, and is orthogonal and complementary to the pipeline model parallelism advocated by approaches such as (Huang et al., 2018). In the remainder of this section, we describe some implementation details of our method.

3.1 Model Parallel Random Number Generation

Techniques that utilize random number generation, such as dropout (Srivastava et al., 2014), are a staple of modern deep learning training. Transformers have dropout layers outside the model parallel regions before residual connections and within model parallel regions in the self attention block. Because some dropout layers are in a model parallel region, while others are not, we need to treat random number generation carefully to ensure dropout works correctly. To synchronize residual connection dropout across model parallel workers we seed the random number generators at the beginning of training with the same seed. This results in identical dropout patterns across all model parallel workers. However, dropout within a model parallel region should result in different random patterns for each worker to achieve randomness across the entire operation. To achieve this we maintain a separate random number generator for dropout within model parallel regions. This random number generator is uniquely seeded for each model parallel worker.

3.2 Hybrid Model and Data Parallelism

Model parallelism is orthogonal to data parallelism, and so we can use both simultaneously to train large models in a reasonable amount of time. Figure 5 shows a grouping of GPUs for hybrid model and data parallelism. Two or more GPUs within the same server form model parallel groups (for example GPUs 1 to 8 in Figure 5), and contain one instance of the model distributed across these GPUs. The remaining GPUs, which could be within the same server but more typically are located in other servers, run additional model parallel groups. GPUs with the same position in each of the model parallel groups (for example GPUs 1, 9, ..., 505 in Figure 5) form data parallel groups so that all GPUs within a data parallel group hold the same model parameters. During back propagation we run multiple gradient all-reduce operations in parallel to reduce weight gradients within each distinct data parallel group. The total number of required GPUs is the product of the number of model and data parallel groups. For example, for the 8.3 billion parameter model we use 8 GPUs per model parallel group and 64-way data parallelism, for a total of 512 GPUs. All communication is implemented in PyTorch by Python calls to NCCL. GPUs within each model parallel group perform all-reduces amongst all GPUs within the group. For data parallelism, each of the all-reduce operations takes place with one of the GPUs from each model parallel group.
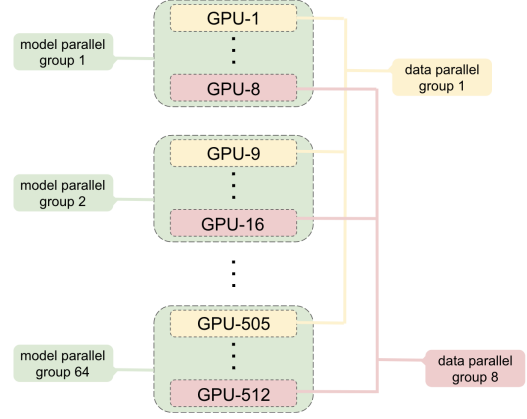

Figure 5. Grouping of GPUs for hybrid model and data parallelism with 8-way model parallel and 64-way data parallel.

4 LANGUAGE MODELING SETUP

Language modeling is a central task in natural language processing and language understanding. There are several formulations of language modeling, but we focus on the most commonly used formulation: for the sequence $X = \{x_1, \cdots, x_m\}$, predict the next token x_t, given the previous tokens. Using cross-entropy loss, we minimize

$$-\sum_i log P_\theta(x_i | x_{i-c}, \cdots, x_{i-1})$$

where c is the context used and P_θ is the conditional probability of the next token.

4.1 Training Language Models

4.1.1 Training Optimization and Hyperparameters

To train our model we follow a procedure largely based on the training procedures described in (Radford et al., 2018; 2019b;a) with a few additions. All training is performed with sequences of 1024 subword units at a batch size of 512 for 300k iterations. To train our models efficiently we utilize mixed precision training with dynamic loss scaling to take advantage of the V100's Tensor Cores (Micikevicius et al., 2017; NVIDIA, 2018). We start by initializing our weights W with a simple normal distribution $W \sim \mathcal{N}(0, 0.02)$. We then scale weights immediately before residual layers by $\frac{1}{\sqrt{2N}}$ where N is the number of transformer layers comprised of self attention and MLP blocks. For our optimizer we utilize an Adam optimizer (Kingma & Ba, 2014) with weight decay (Loshchilov & Hutter, 2019) $\lambda = 0.01$. Additionally, we use global gradient norm clipping of 1.0 to improve the stability of training large models. Our learning rate of 1.5e-4 utilizes a warmup period of 3000 iterations before following a single cycle cosine decay over the remaining 297k iterations. We stop the decay at a minimum learning rate of 1e-5. In our experiments we found that tuning

the learning rate of a particular model via cross-validation improved accuracies, but for the sake of simplicity we consider one learning rate across all model sizes. In all cases, a dropout of 0.1 is used. Lastly, to better manage our memory footprint we utilize activation checkpointing (Chen et al., 2016) after every transformer layer.

4.1.2 *Training Dataset*

To collect a large diverse training set with longterm dependencies we aggregate several of the largest language modeling datasets. We create an aggregate dataset consisting of Wikipedia (Devlin et al., 2018), CC-Stories (Trinh & Le, 2018), RealNews (Zellers et al., 2019), and OpenWebtext. To avoid training set leakage into our downstream tasks we remove the Wikipedia articles present in the WikiText103 test set (Merity et al., 2016). We also remove unnecessary newlines from the CC-Stories corpus introduced by preprocessing artifacts.

For *OpenWebText*, we created a dataset downloaded from *Reddit*, a social media platform, for training. The dataset is conceptually similar to the webtext dataset used in (Radford et al., 2019b). We scraped this dataset with the publicly available OpenWebText codebase[1]. We first scraped all the outgoing URLs with at least 3 karma score from Reddit. We then filtered out URLs with blacklisted domains (e.g. adult content or image and video hosting sites), blacklisted types (e.g. jpg, exe, ppt, etc), and duplicate URLs. We used the *newspaper* library to download the text from each URL then applied *langdetect*[2] to filter out non-english content and *ftfy*[3] to normalize unicode text.

We combined all the datasets and then filtered out all the documents with content length less than 128 tokens from the aggregated dataset. Since similar content might appear multiple times in the aggregated datasets, we used LSH to deduplicate content with a jaccard similarity greater than 0.7. The resulting aggregate corpus contains 174 GB of deduplicated text.

To ensure we do not train on any data found in our test sets, we calculate the percentage of test set 8-grams that also appear in our training set as done in previous work (Radford et al., 2019b). To calculate the overlap we also use a Bloom filter but with a more conservative false positive rate of 10^{-3} to save computation cost. The WikiText103 test set has at most 10.8% overlap and the LAMBADA test set (Paperno et al., 2016) has at most 1.4% overlap. We should note that the WikiText103 test set has already 9.09% overlap with the WikiText103 training set (Radford et al., 2019b). As these are consistent with previous work, we are confident that no

documents from our test data are inadvertently included in our training data.

4.1.3 *Data Loading*

For training, we apply byte-pair encoding tokenization (Bojanowski et al., 2017; Radford et al., 2019b) and randomly split this dataset into a 29:1 ratio to obtain training (168.2 GB) and validation (5.8 GB) sets, respectively. We divide the training set into 5 equal shards and shuffle the processing order of these shards. Within each shard, we shuffle the order of the documents and add an *end of text* token at the end of each document. We then flatten the entire shard and chunk it into 1024 token portions. We then shuffle these chunks randomly one last time before presenting them to the model. We repeat this randomization process every epoch.

4.2 Evaluating Language Models

To analyze the effect increasing model size has on a model's ability to understand language, we need suitable evaluation criteria. Two commonly utilized criteria are language model perplexity on the WikiText103 dataset (Merity et al., 2016) and cloze-style prediction accuracy on the LAMBADA dataset(Paperno et al., 2016).

4.2.1 *Wikitext103 Perplexity*

WikiText103 perplexity is an evaluation criterion that has been well studied over the past few years since the creation of the benchmark dataset. Perplexity is the exponentiation of the average cross entropy of a corpus (Mikolov et al., 2011). This makes it a natural evaluation metric for language models which represent a probability distribution over entire sentences or texts.

$$PPL = \exp(-\frac{1}{T_o} \sum_t^T \log P(t|0:t-1)) \qquad (4)$$

To calculate perplexity in (4) we tokenize the WikiText103 test corpus according to our subword vocabulary and sum the cross entropy loss from each token $[0, T]$. We then normalize the cross entropy loss by the number of tokens in the original tokenization scheme T_o. The WikiText103 test corpus already comes pre-tokenized with word level tokens that prior works have used to compute perplexity. To evaluate our models' perplexities on a level playing field with prior works we must normalize by the original number of tokens, T_o, rather than the number of tokens, T, actually in the tokenized data fed as input to our model. This pre-tokenization also introduces artifacts in the text that are not present in our training data. To alleviate this distributional mismatch, we first preprocess the WikiText103 test dataset with invertible detokenizers to remove various artifacts related to punctuation and whitespace. The value of T_o is calculated before

this preprocessing. For WikiText103's test set $T_o = 245566$ and $T = 270329$.

We must also make one further transformer-specific modification to the perplexity calculation. Unlike RNN-based language models, transformers operate on a fixed window input size. Therefore they cannot fully calculate $P(t|0 : t-1)$ and can only calculate $P(t|t - w : t - 1)$ where w is the size of our context: 1024 tokens. However, calculating this value for every token in our dataset is prohibitively expensive since we must compute approximately T evaluations of a w sized context. To evaluate our models efficiently we take a middle ground approach termed *overlapping evaluation* where we advance the sliding window by some overlap o each time and only compute the cross entropy losses corresponding to the last o tokens of the window. In our experiments we utilize an overlap o of 32, and compute losses over all sliding windows in such a fashion.

4.2.2 LAMBADA Cloze Accuracy

The capability to handle long term contexts is crucial for state of the art language models and is a necessary prerequisite for problems like long-form generation and document-based question answering. Cloze-style datasets like LAMBADA are designed to measure a model's ability to operate in and reason about these types of long term contexts. Cloze-style reading comprehension uses a context of word tokens $x = x_{1:t}$ with one token x_j masked; the models objective is to correctly predict the value of the missing jth token. To accurately predict the missing token, the model requires an in-depth understanding of the surrounding context and how language should be used in such a context. LAMBADA uses cloze-style reading comprehension to test generative left-to-right language models by constructing examples of 4-5 sentences where the last word in the context x_t is masked. Our models utilize subword units, so for LAMBADA evaluation we utilize the raw, unprocessed LAMBADA dataset and require that our model predict the multiple subword tokens that make up the word token. We use teacher forcing, and consider an answer correct only when all output predictions are correct. This formulation is equivalent to the original task of word token prediction.

5 EXPERIMENTS

All of our experiments are conducted on NVIDIA's DGX SuperPod[4] and we use up to 32 DGX-2H servers (a total of 512 Tesla V100 SXM3 32GB GPUs). This system is optimized for multi-node deep learning applications, with 300 GB/sec bandwidth between GPUs inside a server via NVSwitch and 100 GB/sec of interconnect bandwidth be-

---

[4]See https://devblogs.nvidia.com/dgx-superpod-world-record-supercomputing-enterprise/

tween servers using 8 InfiniBand adapters per server.

5.1 Scaling Analysis

To test the scalability of our implementation, we consider GPT-2 models with four sets of parameters detailed in Table 1. To have consistent GEMM sizes in the self attention layer, the hidden size per attention head is kept constant at 96 while the number of heads and layers are varied to obtain configurations ranging from 1 billion to 8 billion parameters. The configuration with 1.2 billion parameters fits on a single GPU whereas the 8 billion parameter model requires 8-way model parallelism (8 GPUs). The original vocabulary size was 50,257, however, to have efficient GEMMs for the logit layer, it is beneficial for the per-GPU vocabulary size to be a multiple of 128. Since we study up to 8-way model parallelism, we pad the vocabulary such that it is divisible by $128 \times 8 = 1024$, resulting in a padded vocabulary size of 51,200. We study both model and model+data parallel scaling. For the model parallel scaling, a fixed batch size of 8 is used across all configurations. Data parallel scaling is necessary for training many state of the art models which typically use a much larger global batch size. To this end, for the model+data parallel cases we fix the global batch size to 512 for all experiments which corresponds to 64-way data parallelism.

Table 1. Parameters used for scaling studies. Hidden size per attention head is kept constant at 96.

Hidden Size	Attention heads	Number of layers	Number of parameters (billions)	Model parallel GPUs	Model +data parallel GPUs
1536	16	40	1.2	1	64
1920	20	54	2.5	2	128
2304	24	64	4.2	4	256
3072	32	72	8.3	8	512

5.1.1 Model and Data Parallelism

Throughout this section, we will showcase weak scaling with respect to the model parameters for both model parallel and model+data parallel cases. Weak scaling is typically done by scaling the batch-size, however, this approach does not address training large models that do not fit on a single GPU and it leads to training convergence degradation for large batch sizes. In contrast, here we use weak scaling to train larger models that were not possible otherwise. The baseline for all the scaling numbers is the first configuration (1.2 billion parameters) in Table 1 running on a single GPU. This is a strong baseline as it achieves 39 TeraFLOPS during the overall training process, which is 30% of the theoretical peak FLOPS for a single GPU in a DGX-2H server.

Figure 6 shows scaling values for both model and model+data parallelism. We observe excellent scaling num-
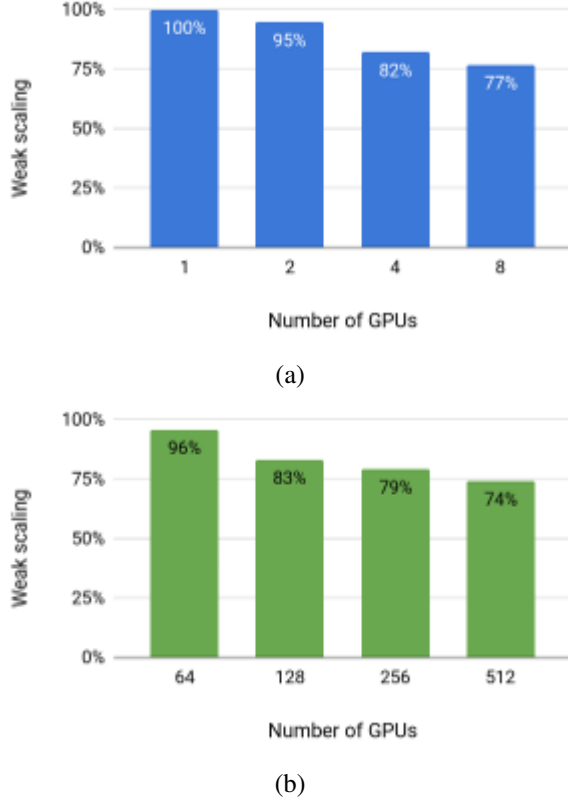
(a)

(b)

Figure 6. Model (a) and model+data (b) parallel weak scaling efficiency as a function of the number of GPUs.

bers in both settings. For example, the 8.3 billion parameters case with 8-way (8 GPU) model parallelism achieves 77% of linear scaling. Model+data parallelism requires further communication of gradients and as a result the scaling numbers drop slightly. However, even for the largest configuration (8.3 billion parameters) running on 512 GPUs, we achieve 74% scaling relative to linear scaling of the strong single GPU baseline configuration (1.2 billion parameters).

5.1.2 *Attention Heads and Scaling*

This section studies the effect of attention heads on model parallel scaling. To this end, we consider the 8.3 billion parameter configuration with 8-way model parallelism and vary the number of heads from 16 to 32. The results are presented in Table 2. As the number of attention heads increases, some of the GEMMS inside the self-attention layer become smaller and also the number of elements in the self attention softmax increases. This results in a slight decrease in scaling efficiency. Future research should be wary of this hyperparameter to design large transformer models that balance model speed and model accuracy.

Table 2. Effect of number of attention heads on scaling on 8.3 billion of parameters with 8-way model parallelism.

Attention heads	Hidden size per head	Scaling Efficiency
16	192	82%
24	128	80%
32	96	77%

Table 3. Speedup obtained for the 1.2 billion parameters model using model parallelism while keeping the batch size constant.

# of GPUs	1	2	4	8
Speedup	1.0	1.64	2.34	2.98

5.1.3 *Strong Scaling*

Our model parallelism is primarily designed to enable training models larger than what can fit in the memory of a single GPU, but it can also accelerate the training of smaller models without increasing the batch size. To measure this acceleration we train a model with a fixed 1.2 billion parameters. We use a fixed batch size of 8 samples per iteration and increase the number of GPUs using model parallelism. The results are listed in Table 3. Using two GPUs makes training 64% faster. Above that we see diminishing returns as the per-GPU computation decreases and the memory bandwidth and communication overheads begin to dominate.

5.2 Language Modeling Results

To demonstrate that large language models can further advance the state of the art, we consider training models of the sizes and configurations listed in Table 4. The 355M model is equivalent in size and configuration of BERT-Large model (Devlin et al., 2018). The 2.5B model is bigger than the previous largest GPT-2 model, and the 8.3B model is larger than any transformer model ever trained, to the best of our knowledge. To train and evaluate our language models we use the procedure described in section 4. Table 4 also lists the time it takes to advance one epoch which is equivalent to 68,507 iterations. For example, for the 8.3B model on 512 GPUs, each epoch takes around two days. Compared to the configurations used for our scaling studies in Table 1, the 2.5B model is the same, the 8.3B model has 24 attention heads instead of 32, and the 355M is much smaller than any seen previously while still using 64 GPUs to train, leading to the much lower time per epoch.

Figure 7 shows validation perpelixity as a function of number of iterations. As the model size increases, the validation perpelixity decreases and reaches a validation perpelixity of 9.27 for the 8.3B model. We report the zero-shot evalution of the trained models on the LAMBADA and WikiText103 datasets in Table 5. We observe the trend that increasing

Table 4. Model configurations used for language modeling.

Parameter Count	Layers	Hidden Size	Attn Heads	Hidden Size per Head	Total GPUs	Time per Epoch (days)
355M	24	1024	16	64	64	0.86
2.5B	54	1920	20	96	128	2.27
8.3B	72	3072	24	128	512	2.10

model size also leads to lower perplexity on WikiText103 and higher cloze accuracy on LAMBADA. Our 8.3B model achieves state of the art perplexity on the WikiText103 test set at a properly adjusted perplexity of 10.81. At 66.51% accuracy, the 8.3B model similarly surpasses prior cloze accuracy results on the LAMBADA task. Several samples generated from the model are provided in the appendix.
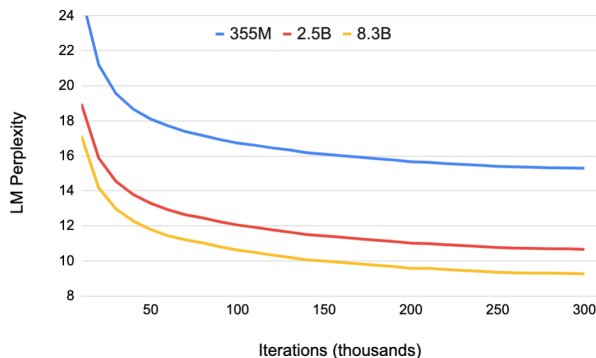

Figure 7. Validation set perplexity. All language models are trained for 300k iterations. Larger language models converge noticeably faster and converge to lower validation perplexities than their smaller counterparts.

Table 5. Zero-shot results. SOTA for Wikitext103 and LAMBADA are from (Krause et al., 2019) and (Radford et al., 2019b), respectively.

Model	Wikitext103 Perplexity ↓	LAMBADA Accuracy ↑
355M	19.31	45.18%
2.5B	12.76	61.73%
8.3B	**10.81**	**66.51%**
SOTA	16.43	63.24%

6 DISCUSSION AND FUTURE WORK

There are several directions for future work. Continuing to increase the scale of pretraining is a promising line of investigation that will further test existing deep learning hardware and software. To realize this, improvements in the efficiency and memory footprint of optimizers will be needed. Scaling

batch size is another approach to improve efficiency of training. However, larger batch sizes require more memory for activations and approaches such as gradient accumulation will be needed. In addition, training a model with more than 16 billion parameters will demand more memory than is available within 16 GPUs of a DGX-2H box. For such models, a hybrid intra-layer and inter-layer model parallelism along with inter-node model parallelism would be more suitable.

Increasing the scale of GPT-2 pretraining and transfer is not the only way to demonstrate the effectiveness of large scale language modeling. To this end three directions of investigation include (a) pretraining different model families (e.g. BERT, Transformer-XL, and XLNet), (b) evaluating performance of large models across more difficult and diverse downstream tasks (e.g. Question Answering, Summarization, and Conversation), and (c) using knowledge distillation to train small student models from our large pretrained teacher models.

7 CONCLUSION

In this work, we trained the world's largest transformer based language model using existing deep learning hardware, software, and models. In doing so, we successfully surpassed the limitations posed by traditional single-GPU-per-model training by implementing a simple and efficient model parallel approach with only a few targeted modifications to the existing PyTorch transformer implementations. We efficiently trained an 8.3 billion parameter language model (24x and 5.6x larger than the size of BERT and GPT-2, respectively) on 512 NVIDIA V100 GPUs with 8-way model parallelism and achieved up to 15.1 PetaFLOPs per second sustained over the entire application. With weak scaling, we found that increasingly large transformer models can be trained in a similar amount of time compared to their smaller counterparts and can demonstrably improve application accuracies. Our larger language models demonstrate this by achieving far superior results on downstream tasks and establish new SOTA for WikiText103 and LAMBADA datasets. Finally, we open sourced our code to enable future work leveraging model parallel transformers and further advance the state of the art on various downstream NLP applications.

REFERENCES

Abadi, M., Agarwal, A., Barham, P., Brevdo, E., Chen, Z., Citro, C., Corrado, G. S., Davis, A., Dean, J., Devin, M., Ghemawat, S., Goodfellow, I., Harp, A., Irving, G., Isard, M., Jia, Y., Jozefowicz, R., Kaiser, L., Kudlur, M., Levenberg, J., Mané, D., Monga, R., Moore, S., Murray, D., Olah, C., Schuster, M., Shlens, J., Steiner, B., Sutskever, I., Talwar, K., Tucker, P., Vanhoucke, V., Vasudevan, V., Viégas, F., Vinyals, O., Warden, P., Wattenberg, M., Wicke, M., Yu, Y., and Zheng, X. TensorFlow: Large-scale machine learning on heterogeneous systems, 2015. URL http://tensorflow.org/. Software available from tensorflow.org.

Ba, J. L., Kiros, J. R., and Hinton, G. E. Layernorm. *CoRR*, abs/1607.06450, 2016. URL http://arxiv.org/abs/1607.06450.

Bojanowski, P., Grave, E., Joulin, A., and Mikolov, T. Enriching word vectors with subword information. *Transactions of the Association for Computational Linguistics*, 5: 135–146, 2017.

Chen, C.-C., Yang, C.-L., and Cheng, H.-Y. Efficient and robust parallel dnn training through model parallelism on multi-gpu platform. *arXiv:1809.02839*, 2018.

Chen, T., Xu, B., Zhang, C., and Guestrin, C. Training deep nets with sublinear memory cost. *CoRR*, abs/1604.06174, 2016. URL http://arxiv.org/abs/1604.06174.

Dai, Z., Yang, Z., Yang, Y., Carbonell, J. G., Le, Q. V., and Salakhutdinov, R. Transformer-xl: Attentive language models beyond a fixed-length context. *CoRR*, abs/1901.02860, 2019. URL http://arxiv.org/abs/1901.02860.

Devlin, J., Chang, M.-W., Lee, K., and Toutanova, K. Bert: Pre-training of deep bidirectional transformers for language understanding, 2018.

Goyal, P., Dollár, P., Girshick, R. B., Noordhuis, P., Wesolowski, L., Kyrola, A., Tulloch, A., Jia, Y., and He, K. Accurate, large minibatch SGD: training imagenet in 1 hour. *CoRR*, abs/1706.02677, 2017.

Harlap, A., Narayanan, D., Phanishayee, A., Seshadri, V., Devanur, N., Ganger, G., and Gibbons, P. Pipedream: Fast and efficient pipeline parallel dnn training. *arXiv:1806.03377*, 2018.

Hendrycks, D. and Gimpel, K. Bridging nonlinearities and stochastic regularizers with gaussian error linear units. *CoRR*, abs/1606.08415, 2016. URL http://arxiv.org/abs/1606.08415.

Hochreiter, S. and Schmidhuber, J. Long short-term memory. *Neural computation*, 9(8):1735–1780, 1997.

Howard, J. and Ruder, S. Fine-tuned language models for text classification. *CoRR*, abs/1801.06146, 2018.

Huang, Y., Cheng, Y., Chen, D., Lee, H., Ngiam, J., Le, Q. V., and Chen, Z. Gpipe: Efficient training of giant neural networks using pipeline parallelism. *CoRR*, abs/1811.06965, 2018. URL http://arxiv.org/abs/1811.06965.

Jia, Z., Zaharia, M., and Aiken, A. Beyond data and model parallelism for deep neural networks. *arXiv:1807.05358*, 2018.

Keskar, N. S., Mudigere, D., Nocedal, J., Smelyanskiy, M., and Tang, P. T. P. On large- batch training for deep learning: Generalization gap and sharp minima. *ICLR*, 2017.

Kingma, D. P. and Ba, J. Adam: A method for stochastic optimization. *arXiv preprint arXiv:1412.6980*, 2014.

Krause, B., Kahembwe, E., Murray, I., and Renals, S. Dynamic evaluation of transformer language models. *arXiv:1904.08378*, 2019.

Li, M., Andersen, D. G., Park, J. W., Smola, A. J., Ahmed, A., Josifovski, V., Long, J., Shekita, E. J., and Su, B.-Y. Scaling distributed machine learning with the parameter server. pp. 583–598, 2014.

Liu, X., He, P., Chen, W., and Gao, J. Multi-task deep neural networks for natural language understanding. *CoRR*, abs/1901.11504, 2019a. URL http://arxiv.org/abs/1901.11504.

Liu, Y., Ott, M., Goyal, N., Du, J., Joshi, M., Chen, D., Levy, O., Lewis, M., Zettlemoyer, L., and Stoyanov, V. Roberta: A robustly optimized BERT pretraining approach. *CoRR*, abs/1907.11692, 2019b. URL http://arxiv.org/abs/1907.11692.

Loshchilov, I. and Hutter, F. Decoupled weight decay regularization. In *International Conference on Learning Representations*, 2019. URL https://openreview.net/forum?id=Bkg6RiCqY7.

McCann, B., Bradbury, J., Xiong, C., and Socher, R. Learned in translation: Contextualized word vectors. *CoRR*, abs/1708.00107, 2017.

Melamud, O., Goldberger, J., and Dagan, I. context2vec: Learning generic context embedding with bidirectional lstm. In *Proceedings of The 20th SIGNLL Conference on Computational Natural Language Learning*, pp. 51–61, 01 2016.

Merity, S., Xiong, C., Bradbury, J., and Socher, R. Pointer sentinel mixture models. *CoRR*, abs/1609.07843, 2016. URL http://arxiv.org/abs/1609.07843.

Micikevicius, P., Narang, S., Alben, J., Diamos, G. F., Elsen, E., Garcia, D., Ginsburg, B., Houston, M., Kuchaiev, O., Venkatesh, G., and Wu, H. Mixed precision training. *CoRR*, abs/1710.03740, 2017.

Mikolov, T., Deoras, A., Kombrink, S., Burget, L., and Černockỳ, J. Empirical evaluation and combination of advanced language modeling techniques. In *Twelfth Annual Conference of the International Speech Communication Association*, 2011.

Mikolov, T., Sutskever, I., Chen, K., Corrado, G., and Dean, J. Distributed representations of words and phrases and their compositionality. *CoRR*, abs/1310.4546, 2013.

NVIDIA. Mixed precision training: Choosing a scaling factor, 2018. URL https://docs.nvidia.com/deeplearning/sdk/mixed-precision-training/index.html#scalefactor.

Paperno, D., Kruszewski, G., Lazaridou, A., Pham, Q. N., Bernardi, R., Pezzelle, S., Baroni, M., Boleda, G., and Fernández, R. The LAMBADA dataset: Word prediction requiring a broad discourse context. *CoRR*, abs/1606.06031, 2016. URL http://arxiv.org/abs/1606.06031.

Pennington, J., Socher, R., and Manning, C. D. Glove: Global vectors for word representation. 2014. URL https://www.aclweb.org/anthology/D14-1162.

Peters, M. E., Neumann, M., Iyyer, M., Gardner, M., Clark, C., Lee, K., and Zettlemoyer, L. Deep contextualized word representations. *CoRR*, abs/1802.05365, 2018. URL http://arxiv.org/abs/1802.05365.

Radford, A., Józefowicz, R., and Sutskever, I. Learning to generate reviews and discovering sentiment. *CoRR*, abs/1704.01444, 2017.

Radford, A., Narasimhan, K., Salimans, T., and Sutskever, I. Improving language understanding by generative pre-training. 2018. URL https://blog.openai.com/language-unsupervised/.

Radford, A., Child, R., Gray, S., and Sutskever, I. Generating long sequences with sparse transformers. 2019a. URL https://openai.com/blog/sparse-transformer/.

Radford, A., Wu, J., Child, R., Luan, D., Amodei, D., and Sutskever, I. Better language models and their implications. 2019b. URL https://openai.com/blog/better-language-models/.

Ramachandran, P., Liu, P. J., and Le, Q. V. Unsupervised pretraining for sequence to sequence learning. *CoRR*, abs/1611.02683, 2016. URL http://arxiv.org/abs/1611.02683.

Shazeer, N., Cheng, Y., Parmar, N., Tran, D., Vaswani, A., Koanantakool, P., Hawkins, P., Lee, H., Hong, M., Young, C., Sepassi, R., and Hechtman, B. Mesh-TensorFlow: Deep learning for supercomputers. In *Neural Information Processing Systems*, 2018.

Srivastava, N., Hinton, G., Krizhevsky, A., Sutskever, I., and Salakhutdinov, R. Dropout: A simple way to prevent neural networks from overfitting. *Journal of Machine Learning Research*, 15:1929–1958, 2014. URL http://jmlr.org/papers/v15/srivastava14a.html.

Trinh, T. H. and Le, Q. V. A simple method for common-sense reasoning. *CoRR*, abs/1806.02847, 2018. URL http://arxiv.org/abs/1806.02847.

Turian, J., Ratinov, L., and Bengio, Y. Word representations: A simple and general method for semi-supervised learning. In *Proceedings of the 48th Annual Meeting of the Association for Computational Linguistics*, ACL '10, pp. 384–394, Stroudsburg, PA, USA, 2010. Association for Computational Linguistics.

Valiant, L. G. A bridging model for parallel computation. *Communications of the ACM*, 33(8):103-111, 1990.

Vaswani, A., Shazeer, N., Parmar, N., Uszkoreit, J., Jones, L., Gomez, A. N., Kaiser, L., and Polosukhin, I. Attention is all you need. *CoRR*, abs/1706.03762, 2017.

Yang, Z., Dai, Z., Yang, Y., Carbonell, J. G., Salakhutdinov, R., and Le, Q. V. Xlnet: Generalized autoregressive pretraining for language understanding. *CoRR*, abs/1906.08237, 2019. URL http://arxiv.org/abs/1906.08237.

You, Y., Gitman, I., and Ginsburg, B. Large batch training of convolutional networks. *arXiv:1708.03888*, 2017.

You, Y., Li, J., Reddi, S., Hseu, J., Kumar, S., Bhojanapalli, S., Song, X., Demmel, J., and Hsieh, C.-J. Large batch optimization for deep learning: Training bert in 76 minutes. *arXiv:1904.00962*, 2019.

Zellers, R., Holtzman, A., Rashkin, H., Bisk, Y., Farhadi, A., Roesner, F., and Choi, Y. Defending against neural fake news. *CoRR*, abs/1905.12616, 2019. URL http://arxiv.org/abs/1905.12616.

A TEXT SAMPLES

Below are some text samples generated by Megatron-LM using a context prompt. Some of the texts are cut short.

Context: There is a major force uniting America's fiercely partisan politicians: big technology companies. Democrats and Republicans at the federal and state levels are coming together to scrutinize the power of the Silicon Valley giants and, potentially, to rein them in. Letitia James, the Democratic attorney general of New York, announced on Friday that attorneys general in eight states — four Democrats and four Republicans — and the District of Columbia had begun an antitrust investigation of Facebook.

Megatron-LM: The probe, which Ms. James said was likely to result in a lawsuit, is the latest in a string of high-profile regulatory actions, in state and federal courts, that challenge the market dominance of tech companies. The Justice Department is also investigating Facebook's data practices and whether the company violated a 2011 consent decree with the Federal Trade Commission that required the company to obtain explicit consent from users before sharing their personal information with third parties.

What might prompt a bipartisan coalition to sue Facebook? It isn't clear that there's any one thing. But lawmakers and regulators have been increasingly alarmed by the company's recent controversies, which include a data breach affecting more than 50 million users, the spread of fake news on the platform, and the spreading of propaganda by Russian-backed accounts during the 2016 election.

It's also not clear how serious the Democratic attorneys general are about filing a lawsuit. While the attorneys general announced their investigation in a press release, they said they had not yet reached a decision on whether to sue. In the past, some of the attorneys general have focused their efforts on trying to extract settlements from the tech companies without filing lawsuits.

"As attorneys general who have investigated and prosecuted health-care fraud, price-fixing, and other violations of our states' antitrust laws, we are committed to holding companies accountable when they fail to protect consumers and competition in the market," Ms. James said in a statement.

The investigation, which was first reported by The New York Times, was announced as the companies that dominate the tech industry are facing scrutiny from antitrust regulators around the world. Google, Twitter, Facebook and Amazon have faced a series of antitrust probes and class-action lawsuits in the United States and Europe ...

Context: A flower, sometimes known as a bloom or blossom, is the reproductive structure found in flowering plants (plants of the division Magnoliophyta, also called angiosperms). The biological function of a flower is to effect reproduction, usually by providing a mechanism for the union of sperm with eggs.

Megatron-LM: The structural components of the flower are the petals and sepals, which are the reproductive organs, the hypanthium, which forms the base of the flower, and the calyx, which is the stigma. A flower is a composite structure, made up of sepals and petals, the sepals being composed of three fused tepals and the petals of two fused petals, though there are a few exceptions to this rule. The three sepals are fused to form a tube called the calyx, which is the structure that attracts the pollinating insects. The petals are composed of two fused sepals (the outer ones) which open and fold inward (clasp) at the tips, forming the petals. The four sepals and petals are united to form a corolla, the outermost structure of a flower that carries pollen.

The structure of the flower has important functional significance, including protection of the ovary from insects and the transfer of pollen between the male and female parts of a flower. The structure of the flower is also used by the plants to attract pollinators, for example by the shape of the petals and the colours of the stamens.

The structure of a flower varies widely between plants. The simplest flowers are unbranched and made up of only two sepals and a single petal. Other examples are composed of several petals joined together, with the innermost sepals completely fused to form a tube called the calyx. The petals are usually coloured to attract insects, which can only see certain wavelengths of light. The sepals are often coloured too, but usually have a more limited range of colours.

Some flowers have additional structures, such as stamens, anthers and ovaries. The presence of stamens and anthers in some flowers is used for cross-pollination. The ovary is the organ where the female gametes are produced, and is the first structure that is developed in an embryo of a flower. The ovary is the female flower's reproductive system, and consists of the ovary wall, the ovule, and the oviduct.

Flowers vary in their size and shape, from tiny to huge, and are often brightly coloured. Flowers may be seen as part of a larger structure, such as a flower head, an inflorescence, a bract, a spathe, or a carpels.

The term "flower" is usually applied to the floral structures that are visible externally, but many ...

Context: OpenAI did not release the full GPT-2 model due to concerns of malicious use, but they did release a smaller version equivalent in size to the original GPT (117 M parameters), trained on the new, larger dataset. Although not as powerful as the large model, the smaller version still has some language generation chops.

Megatron-LM: In December 2016, DeepMind released an even larger model, GPT-3, with more than 1.5 billion parameters. DeepMind has no plans to release the model, so the question remains whether other AI researchers will be able to copy and build on it.

Other language models have been built using other techniques. In 2016, a team of researchers at Columbia University, the University of Toronto, and Google DeepMind showed that neural network-based models, including deep neural networks, can learn the grammatical rules of a language. A group at Google DeepMind demonstrated that these rules could be represented in a network as a kind of decision tree, where the nodes in the tree are the words in the sentence, and the rules for the words are represented by the decision nodes at the end of the tree. The network would go through the network in order to choose the next word in the sentence based on whether it follows the rules or not. In 2017, the team at DeepMind released a paper detailing the training and performance of their model.

The problems with the model are many. One of the biggest issues is that grammatical rules are ambiguous. For example, the phrase "I came down" could mean I came down from the stairs, or I came down from the sky. In English, there are numerous other examples of sentences that are ambiguous. The model does not have any way to tell the difference between the ambiguous sentences, and so it simply makes a prediction about the entire sentence based on what rules it does know. This makes the model vulnerable to the phenomenon of overfitting. In order to avoid this, models must be trained using a large set of data, so that the model can find useful patterns in the data, and use those patterns to help make predictions. This approach is also prone to errors, as it relies on the training data to help make correct predictions, and the data might be incorrectly labeled, resulting in a model that produces a poor set of predictions. In the case of GPT-2, the model only has about 100 training sentences for each type of phrase (subject-verb-object).

Additionally, languages have different rules for different types of phrasing. In English, the subject-verb-object construction is fairly universal, but other languages have different rules, which makes training a GPT more complicated.

Finally, there is an inherent trade-off between the number of rules and the performance of the model ...

Context: The 36th International Conference on Machine Learning (ICML 2019) will be held in Long Beach, CA, USA from June 10th to June 15th, 2019. The conference will consist of one day of tutorials (June 10), followed by three days of main conference sessions (June 11-13), followed by two days of workshops (June 14-15).

Megatron-LM: With a broad scope, the conference addresses the challenges and opportunities in machine learning for practitioners and researchers. The conference program will feature sessions that cover different machine learning topics, from reinforcement learning to computational biology and from AI to deep learning. It is the world's premier forum for exploring and presenting advances in machine learning.

ICML 2019 will offer an impressive range of sessions, which will be of interest to practitioners, researchers, and students. Some of the topics we will cover include:

Reinforcement learning

Soft Computing

Computational biology

Convolutional networks

Deep learning

Statistical learning theory

Statistical learning machine

Natural language processing

Biology

Awards

Since its first edition, the conference has grown and evolved. Today, it is the premier forum for showcasing state-of-the-art research in machine learning. The conference is widely recognized as the preeminent venue for showcasing the latest research in machine learning, and a place where practitioners, researchers, and students can exchange ideas and insights.

The conference is a peer-reviewed venue that draws its prestige and influence from the excellence of the presentations and the quality of the invited speakers. With over 50% of all talks selected based on their content by an expert committee, it is clear that the conference's high standards of research and quality keep a very high bar.

Context: Rescue workers on Tuesday found a flight recorder from the Russian plane that crashed into the Black Sea over the weekend, the defense ministry said. All 92 people aboard the Russian military's Tu-154 plane are believed to have died Sunday morning when it crashed two minutes after taking off from the southern Russian city of Sochi.

Megatron-LM: A child holds up a portrait of a dead relative at a makeshift morgue in a building belonging to a local government in the town of Petropavlovsk-Kamchatsky. A Russian Defense Ministry Tu-154 plane with 92 people on board crashed into the Black Sea on Sunday morning shortly after taking off from the southern Russian city of Sochi. (Yuri Kochetkov/EPA)

According to the ministry, the plane was found at a depth of more than 1,000 feet and the recorder was damaged. "The defense ministry has decided to hand over the recorder to the Interstate Aviation Committee, which is the main investigative body in the air crash," the statement said.

The search for the plane's black boxes is being hampered by strong winds and rough seas. Investigators have launched a criminal probe into the crash, which officials said was likely caused by a technical fault.

Russian President Vladimir Putin declared Wednesday a day of mourning for the victims.