Deep Learning — Assignment 2

Second assignment for the 2020 Deep Learning course (NWI-IMC058) of the Radboud University.

Twan van Laarhoven (tvanlaarhoven@cs.ru.nl) and Gijs van Tulder (g.vantulder@cs.ru.nl)

September 2020

Names: Eugenia Martynova (s1038931) & Denise Meerkerk (s4467132)

Group: 3

Instructions:

- Fill in your names and the name of your group.
- Answer the questions and complete the code where necessary.
- Re-run the whole notebook before you submit your work.
- Save the notebook as a PDF and submit that in Brightspace together with the .ipynb notebook file.
- The easiest way to make a PDF of your notebook is via File > Print Preview and then use your browser's print option to print to PDF.

Objectives¶

In this assignment you will

- 1. Learn how to define and train a neural network with pytorch
- 2. Experiment with convolutional neural networks
- 3. Investigate the effect of dropout and batch normalization

Required software¶

If you haven't done so already, you will need to install the following additional libraries:

- · torch and torchvision for PyTorch,
- d21, the library that comes with Dive into deep learning book,
- sounddevice to play audio,
- python_speech_features to compute MFCC features.

All libraries can be installed with pip install.

In [6]:

pip install d2l

Requirement already satisfied: d2l in /home/denise/anaconda3/lib/python3.7/site-packages (0.14.3)

Requirement already satisfied: pandas in /home/denise/anaconda3/lib/python3.7/site-packages (from d2l) (1.0.3)

```
Requirement already satisfied: jupyter in /home/denise/anaconda3/lib/python3.7/site-
packages (from d2l) (1.0.0)
Requirement already satisfied: numpy in /home/denise/anaconda3/lib/python3.7/site-
packages (from d2l) (1.18.1)
Requirement already satisfied: matplotlib in
/home/denise/anaconda3/lib/python3.7/site-packages (from d2l) (3.1.3)
Requirement already satisfied: pytz>=2017.2 in /home/denise/anaconda3/lib/python3.7/
site-packages (from pandas->d2l) (2019.3)
Requirement already satisfied: python-dateutil>=2.6.1 in /home/denise/anaconda3/lib/
python3.7/site-packages (from pandas->d2l) (2.8.1)
Requirement already satisfied: nbconvert in
/home/denise/anaconda3/lib/python3.7/site-packages (from jupyter->d2l) (5.6.1)
Requirement already satisfied: qtconsole in
/home/denise/anaconda3/lib/python3.7/site-packages (from jupyter->d2l) (4.7.2)
Requirement already satisfied: ipywidgets in
/home/denise/anaconda3/lib/python3.7/site-packages (from jupyter->d2l) (7.5.1)
Requirement already satisfied: ipykernel in
/home/denise/anaconda3/lib/python3.7/site-packages (from jupyter->d2l) (5.1.4)
Requirement already satisfied: notebook in
/home/denise/anaconda3/lib/python3.7/site-packages (from jupyter->d2l) (6.0.3)
Requirement already satisfied: jupyter-console in
/home/denise/anaconda3/lib/python3.7/site-packages (from jupyter->d2l) (6.1.0)
Requirement already satisfied: cycler>=0.10 in /home/denise/anaconda3/lib/python3.7/
site-packages (from matplotlib->d2l) (0.10.0)
Requirement already satisfied: kiwisolver>=1.0.1 in
/home/denise/anaconda3/lib/python3.7/site-packages (from matplotlib->d2l) (1.1.0)
Requirement already satisfied: pyparsing!=2.0.4,!=2.1.2,!=2.1.6,>=2.0.1 in
/home/denise/anaconda3/lib/python3.7/site-packages (from matplotlib->d2l) (2.4.6)
Requirement already satisfied: six>=1.5 in
/home/denise/anaconda3/lib/python3.7/site-packages (from python-dateutil>=2.6.1-
>pandas->d2l) (1.14.0)
Requirement already satisfied: jinja2>=2.4 in
/home/denise/anaconda3/lib/python3.7/site-packages (from nbconvert->jupyter->d2l)
(2.11.1)
Requirement already satisfied: traitlets>=4.2 in
/home/denise/anaconda3/lib/python3.7/site-packages (from nbconvert->jupyter->d2l)
(4.3.3)
Requirement already satisfied: defusedxml in
/home/denise/anaconda3/lib/python3.7/site-packages (from nbconvert->jupyter->d2l)
(0.6.0)
Requirement already satisfied: pandocfilters>=1.4.1 in
/home/denise/anaconda3/lib/python3.7/site-packages (from nbconvert->jupyter->d2l)
(1.4.2)
Requirement already satisfied: entrypoints>=0.2.2 in
/home/denise/anaconda3/lib/python3.7/site-packages (from nbconvert->jupyter->d2l)
Requirement already satisfied: bleach in /home/denise/anaconda3/lib/python3.7/site-
```

```
packages (from nbconvert->jupyter->d2l) (3.1.0)
Requirement already satisfied: nbformat>=4.4 in
/home/denise/anaconda3/lib/python3.7/site-packages (from nbconvert->jupyter->d2l)
(5.0.4)
Requirement already satisfied: jupyter-core in /home/denise/anaconda3/lib/python3.7/
site-packages (from nbconvert->jupyter->d2l) (4.6.3)
Requirement already satisfied: testpath in
/home/denise/anaconda3/lib/python3.7/site-packages (from nbconvert->jupyter->d2l)
(0.4.4)
Requirement already satisfied: mistune<2,>=0.8.1 in
/home/denise/anaconda3/lib/python3.7/site-packages (from nbconvert->jupyter->d2l)
(0.8.4)
Requirement already satisfied: pygments in
/home/denise/anaconda3/lib/python3.7/site-packages (from nbconvert->jupyter->d2l)
(2.6.1)
Requirement already satisfied: qtpy in /home/denise/anaconda3/lib/python3.7/site-
packages (from qtconsole->jupyter->d2l) (1.9.0)
Requirement already satisfied: pyzmq>=17.1 in
/home/denise/anaconda3/lib/python3.7/site-packages (from qtconsole->jupyter->d2l)
(18.1.1)
Requirement already satisfied: jupyter-client>=4.1 in
/home/denise/anaconda3/lib/python3.7/site-packages (from qtconsole->jupyter->d2l)
(6.1.2)
Requirement already satisfied: ipython-genutils in
/home/denise/anaconda3/lib/python3.7/site-packages (from qtconsole->jupyter->d2l)
(0.2.0)
Requirement already satisfied: widgetsnbextension~=3.5.0 in
/home/denise/anaconda3/lib/python3.7/site-packages (from ipywidgets->jupyter->d2l)
(3.5.1)
Requirement already satisfied: ipython>=4.0.0; python version >= "3.3" in
/home/denise/anaconda3/lib/python3.7/site-packages (from ipywidgets->jupyter->d2l)
(7.13.0)
Requirement already satisfied: tornado>=4.2 in /home/denise/anaconda3/lib/python3.7/
site-packages (from ipykernel->jupyter->d2l) (6.0.4)
Requirement already satisfied: prometheus-client in
/home/denise/anaconda3/lib/python3.7/site-packages (from notebook->jupyter->d2l)
(0.7.1)
Requirement already satisfied: Send2Trash in
/home/denise/anaconda3/lib/python3.7/site-packages (from notebook->jupyter->d2l)
(1.5.0)
Requirement already satisfied: terminado>=0.8.1 in
/home/denise/anaconda3/lib/python3.7/site-packages (from notebook->jupyter->d2l)
(0.8.3)
Requirement already satisfied: prompt-toolkit!=3.0.0,!=3.0.1,<3.1.0,>=2.0.0 in
/home/denise/anaconda3/lib/python3.7/site-packages (from jupyter-console->jupyter-
>d21) (3.0.4)
Requirement already satisfied: setuptools in
```

```
/home/denise/anaconda3/lib/python3.7/site-packages (from kiwisolver>=1.0.1-
>matplotlib->d2l) (46.1.3.post20200330)
Requirement already satisfied: MarkupSafe>=0.23 in
/home/denise/anaconda3/lib/python3.7/site-packages (from jinja2>=2.4->nbconvert-
>jupyter->d2l) (1.1.1)
Requirement already satisfied: decorator in
/home/denise/anaconda3/lib/python3.7/site-packages (from traitlets>=4.2->nbconvert-
>jupyter->d2l) (4.4.2)
Requirement already satisfied: webencodings in /home/denise/anaconda3/lib/python3.7/
site-packages (from bleach->nbconvert->jupyter->d2l) (0.5.1)
Requirement already satisfied: jsonschema!=2.5.0,>=2.4 in
/home/denise/anaconda3/lib/python3.7/site-packages (from nbformat>=4.4->nbconvert-
>jupyter->d2l) (3.2.0)
Requirement already satisfied: pexpect; sys platform != "win32" in
/home/denise/anaconda3/lib/python3.7/site-packages (from ipython>=4.0.0;
python version >= "3.3"->ipywidgets->jupyter->d2l) (4.8.0)
Requirement already satisfied: jedi>=0.10 in
/home/denise/anaconda3/lib/python3.7/site-packages (from ipython>=4.0.0;
python version >= "3.3"->ipywidgets->jupyter->d2l) (0.15.2)
Requirement already satisfied: backcall in
/home/denise/anaconda3/lib/python3.7/site-packages (from ipython>=4.0.0;
python version >= "3.3"->ipywidgets->jupyter->d2l) (0.1.0)
Requirement already satisfied: pickleshare in
/home/denise/anaconda3/lib/python3.7/site-packages (from ipython>=4.0.0;
python version >= "3.3"->ipywidgets->jupyter->d2l) (0.7.5)
Requirement already satisfied: wcwidth in /home/denise/anaconda3/lib/python3.7/site-
packages (from prompt-toolkit!=3.0.0,!=3.0.1,<3.1.0,>=2.0.0->jupyter-console-
>jupyter->d2l) (0.1.9)
Requirement already satisfied: attrs>=17.4.0 in
/home/denise/anaconda3/lib/python3.7/site-packages (from jsonschema!=2.5.0,>=2.4-
>nbformat>=4.4->nbconvert->jupyter->d2l) (19.3.0)
Requirement already satisfied: importlib-metadata; python version < "3.8" in
/home/denise/anaconda3/lib/python3.7/site-packages (from jsonschema!=2.5.0,>=2.4-
>nbformat>=4.4->nbconvert->jupyter->d2l) (1.5.0)
Requirement already satisfied: pyrsistent>=0.14.0 in
/home/denise/anaconda3/lib/python3.7/site-packages (from jsonschema!=2.5.0,>=2.4-
>nbformat>=4.4->nbconvert->jupyter->d2l) (0.16.0)
Requirement already satisfied: ptyprocess>=0.5 in
/home/denise/anaconda3/lib/python3.7/site-packages (from pexpect; sys platform !=
"win32"->ipython>=4.0.0; python version >= "3.3"->ipywidgets->jupyter->d2l) (0.6.0)
Requirement already satisfied: parso>=0.5.2 in /home/denise/anaconda3/lib/python3.7/
site-packages (from jedi>=0.10->ipython>=4.0.0; python version >= "3.3"->ipywidgets-
>jupyter->d2l) (0.5.2)
Requirement already satisfied: zipp>=0.5 in
/home/denise/anaconda3/lib/python3.7/site-packages (from importlib-metadata;
python version < "3.8"->jsonschema!=2.5.0,>=2.4->nbformat>=4.4->nbconvert->jupyter-
>d2l) (2.2.0)
```

```
Note: you may need to restart the kernel to use updated packages.
In [7]:
%matplotlib inline
import os
import numpy as np
from d2l import torch as d2l
import torch
from torch import nn
from scipy.io import wavfile
In [8]:
# This is to launch and use terminal from google colab
!pip install kora
from kora import console
console.start()
1 1 1
Out[8]:
'\n!pip install kora\nfrom kora import console\nconsole.start() \n'
```

2.1 Digits dataset¶

The d2l book uses a dataset of images as a running example (FashionMNIST). In this assignment we will investigate CNNs in a completely different domain: speech recognition.

The dataset we use is the free spoken digits dataset, which can be found on https://github.com/Jakobovski/free-spoken-digit-dataset. This dataset consists of the digits 0 to 9, spoken by different speakers. The data comes as .wav files.

Use git clone to download the dataset.

use command below in terminal (the fourth item below files on the left of the screen)

```
git clone https://github.com/Jakobovski/free-spoken-digit-dataset
```

Below is a function to load the data. We pad/truncate each sample to the same length. The raw audio is usually stored in 16 bit integers, with a range -32768 to 32767, where 0 represents no signal. Before using the data, it should be normalized. A common approach is to make sure that the data is between 0 and 1 or between -1 and 1.

Update the below code to normalize the data to a reasonable range

```
In [9]:
samplerate = 8000
def load_waveform(file, size = 6000):
    samplerate, waveform = wavfile.read(file)
    # Take first 6000 samples from waveform. With a samplerate of 8000 that
corresponds to 3/4 second
```

```
# Pad with 0s if the file is shorter
waveform = np.pad(waveform,(0,size))[0:size]
# Normalize waveform between -1 and 1.
waveform = waveform/32768
return waveform
```

The following code loads all .wav files in a directory, and makes it available in a pytorch dataset.

Load the data into a variable data

```
In [10]:
class SpokenDigits(torch.utils.data.Dataset):
    def init (self, data dir):
        digits x = []
        digits y = []
        for file in os.listdir(data dir):
            if file.endswith(".wav"):
                waveform = load waveform(os.path.join(data dir, file))
                label = int(file[0])
                digits x.append(waveform)
                digits_y.append(label)
        self.x = torch.from numpy(np.array(digits x, dtype=np.float32))
        self.x = self.x.unsqueeze(1) # One channel
        self.y = torch.from numpy(np.array(digits y))
    def len (self):
        return len(self.x)
    def getitem (self, idx):
        return self.x[idx], self.y[idx]
data path = "free-spoken-digit-dataset/recordings"
data = SpokenDigits(data path)
In [11]:
# Since SpokenDigits returns the number of samples in __len__ method, we can just
use len(data) to
# obtain this number
print(len(data))
3000
In [12]:
# To check a sample dimensionality we use the size() method of the first sample
print(data.x[0].size())
torch.Size([1, 6000])
In [13]:
```

```
# To obtain number of classes, we check the number of unique items in data.y array
print(len(np.unique(data.y.numpy())))
```

10

Describe the dataset: how many samples are there, what is their dimensionality? How many classes are there?

There are 3000 samples of 1x6000 dimensionality and 10 classes.

Here is code to play samples from the dataset to give you an idea what it "looks" like.

this did not work for us, that is why it's commented.

```
In [14]:
. . .
import sounddevice as sd
def play(sample):
    sd.play(sample[0][0], samplerate)
    print(sample[1])
play(data[0])
1 1 1
Out[14]:
'\nimport sounddevice as sd\ndef play(sample):\n
                                                      sd.play(sample[0][0],
                 print(sample[1])\nplay(data[0])\n'
samplerate)\n
In [15]:
train prop = 2/3
train count = int(len(data) * train prop)
train, test = torch.utils.data.random split(data, [train count, len(data)-
train count])
```

The code above is code to split the data into a training and test set. It uses 2/3 of the data for training.

Discuss an advantage and disadvantage of using more of the data for training

Increase of training data share might help to increase model accuracy since we provide a network with more information about the data. In other words, a model just has more samples to learn from. However, with this approach, we risk missing overfitting because if the test set is too small, there is a larger chance of not having a representative test set. Then, a training set might contain all the "difficult" samples and samples from a test set would be easy to classify. So, we won't be able to evaluate the generalisation properties of the model.

On the contrary, if a test set share is too high, we will estimate the generalization abilities of the model well. However, the network accuracy would be smaller than we could potentially get with the data.

Finally, we split the data into batches:

```
In [16]:
data_params = {'batch_size': 32}
train_iter = torch.utils.data.DataLoader(train, **data_params)
```

2.2 One dimensional convolutional neural network¶

We will now define a network architecture. We will use a combination of convolutional layers and pooling. Note that we use 1d convolution and pooling here, instead of the 2d operations used for images.

Complete the network architecture, look at the d2l book chapters 6 and 7 for examples

```
In [17]:
net = torch.nn.Sequential(
    nn.Convld(1, 4, kernel_size=5), nn.ReLU(),
    nn.AvgPoolld(kernel_size=2, stride=2),
    nn.Convld(4, 8, kernel_size=5), nn.ReLU(),
    nn.AvgPoolld(kernel_size=2, stride=2),
    nn.Convld(8, 16, kernel_size=5), nn.ReLU(),
    nn.AvgPoolld(kernel_size=2, stride=2),
    nn.Convld(16, 32, kernel_size=5), nn.ReLU(),
    nn.AvgPoolld(kernel_size=2, stride=2),
    nn.Flatten(),
    nn.Flatten(),
    nn.Linear(11872, 128), nn.ReLU(),
    nn.Linear(64, 10))
```

The first fully connected layer has input dimension 11872, where does that number come from?

The dimensionality of the network input is 1×000 . Each layer changes the dimentionality of the data. All the convolutional layers in our network use valid padding and stride 1, therefore the dimensionality of their output would be 1×000 where 1×000 where 1×000 where 1×000 is the input length and 1×000 is the kernel size. Also after each convolution layers the number of channels is changed. All the pooling layers have kernel size 2 and stride 2, hence the dimentionality of their output would be 1×000 rfloors.

We will use the notation $(c\times n_m)$ as input/output size where $c\times s$ is the number of channels, $n_m - s$ - first data dimension, $n_m - s$ - second data dimension. Here are the changes which occur in our network step by step:

- 2. AvgPool1d. Input \$4\times1\times5996\$, output \$4\times1\times \lfloor 5996/2 \rfloor = 4\times1\times2998\$
- 3. Conv1d. Input \$4\times1\times2998\$, output \$8\times1\times2994\$
- 4. AvgPool1d. Input \$8\times1\times2994\$, output \$8\times1\times1497\$
- 5. Conv1d. Input \$8\times1\times1497\$, output \$16\times1\times1493\$
- 6. AvgPool1d. Input \$16\times1\times1493\$, output \$16\times1\times746\$
- 7. Conv1d. Input \$16\times1\times746\$, output \$32\times1\times742\$

8. AvgPool1d. Input - \$32\times1\times742\$, output \$32\times1\times371\$

Then, we use Flatten operation which transforms a tensor of size $(c\times n_h\times n_w)$ to a 1-d vector \$1 \times n\$. The size of the previous layer output is \$32\times371\$, hence \$n = 32\times 371 = 11872\$

How many parameters are there in the model? I.e. the total number of weights and biases

```
In [18]:
param_num = sum([param.nelement() for param in net.parameters()])
print(param_num)
1532090
```

Suppose that instead of using convolutions, we had used only fully connected layers. How many parameters would be needed in that case approximately?

A fully connected layer consists of weighs \$\mathbf{W}\$ and biases \$\mathbf{b}\$. The dimensionality of \$\mathbf{W}\$ depend on input and output layer sizes: for \$N\$ inputs and \$H\$ outputs in \$N\times H\$. The dimensionality of \$\mathbf{b}\$ depends on output size and is \$1\times H\$. Hence the number of parameters in a fully connected layer is \$(N+1)\times H\$.

The number of parameters in 3 last fully connected layers in our network is:

```
1. (11872 + 1)\times 128 = 1519744
```

- 2. $(128 + 1)\times 64 = 8256$
- 3. $(64 +1)\times 10 = 650$ \$

Which gives 1528650 parameters.

If we replace 4 convolution layers with fully connected layers with the same number of hidden units \$H\$, we would get the following number of parameters:

```
((((6000 + 1)\times H + 1
```

Even for H=100 it gives about 100 million additional parameters, whereas convolution layers only give 1532090 - 1528650 = 3440 additional parameters.

The FashionMNIST dataset used in the book has 60000 training examples. How large is our training set? How would the difference affect the number of epochs that we need? Compare to chapter 6.6 and 7.1 of the book.

How many epochs do you think are needed?

Out training set has 2000 samples (since we have 3000 samples in total and leave 2/3 in the training set). So, our training set is 30 times smaller than FashionMNIST. Since we have less training data, we will definitely need more epoch for training convergence. But it is difficult to give an estimate of how many since we don't know which other properties of data affect training convergence. Also in our case, the share of the test set is larger, in FashionMNIST training set size is 6/7. A naive way to guess would be to say that we need about 30 times more epochs, so 300. In practice, we found that with a proper LR about 70 epochs is enough for the initial network.

```
In [19]:
```

```
lr, num_epochs = 0.07, 100
```

We will use the code from the d2l book to train the network. In particular, the train_ch6 function,

defined in chapter 6.6. This function is available in the d21 library. However, this function has a bug: it only initializes the weights for 2d convolutional layers, not for 1d convolutional layers.

Make a copy of the train_ch6 function, and correct the error

```
In [20]:
def evaluate accuracy gpu(net, data iter, device=None): #@save
    """Compute the accuracy for a model on a dataset using a GPU."""
    net.eval() # Set the model to evaluation mode
    if not device:
        device = next(iter(net.parameters())).device
   # No. of correct predictions, no. of predictions
   metric = d2l.Accumulator(2)
    for X, y in data_iter:
        X, y = X.to(device), y.to(device)
        metric.add(d2l.accuracy(net(X), y), d2l.size(y))
    return metric[0] / metric[1]
In [21]:
def train(net, train iter, test iter, num epochs, lr, device=d2l.try gpu()):
    """Train a model with a GPU (defined in Chapter 6)."""
    def init weights(m):
        if type(m) == nn.Linear or type(m) == nn.Conv1d:
            torch.nn.init.xavier uniform (m.weight)
    net.apply(init weights)
    print('training on', device)
    net.to(device)
    optimizer = torch.optim.SGD(net.parameters(), lr=lr)
    loss = nn.CrossEntropyLoss()
    animator = d2l.Animator(xlabel='epoch', xlim=[0, num epochs],
                            legend=['train loss', 'train acc', 'test acc'])
   timer = d2l.Timer()
    for epoch in range(num epochs):
        # Sum of training loss, sum of training accuracy, no. of examples
        metric = d2l.Accumulator(3)
        for i, (X, y) in enumerate(train iter):
            timer.start()
            net.train()
            optimizer.zero grad()
            X, y = X.to(device), y.to(device)
            y hat = net(X)
            l = loss(y hat, y)
            l.backward()
            optimizer.step()
            with torch.no grad():
                metric.add(l * X.shape[0], d2l.accuracy(y_hat, y), X.shape[0])
            timer.stop()
            train loss = metric[0]/metric[2]
```

Now train the network.

```
In [22]:
train(net, train_iter, test_iter, num_epochs, lr)
loss 0.000, train acc 1.000, test acc 0.544
963.7 examples/sec on cpu
```

Is the training converged?

If the training has not converged, maybe you need to change the number of epochs and/or the learning rate.

Document the changes that you made and their effect:

At first, we did not increase epoch numbers significantly (probably we started from 20) because we wanted to speed up the choice of learning rate by comparison of how learning curves change at the beginning of training. But with different learning rates in 20 epochs, there was no change. We increased the number of epochs to something close to 100. Then, we tried to decrease the LR to typical values of $0.005\ 0.001$, then increase to $0.05\ and\ 0.1$. We didn't get convergence with any of these values but with higher learning rates ($0.05\ and\ 0.1$) accuracy started to increase (and loss to decrease), however then it diverged again. So we tried different values in between. We found that $0.7,\ 0.75\ and\ 0.8\ LR$ work. In the end, we chose LR = $0.7\ and\ adjusted$ the number of epochs according to convergence.

It should be noticed that training is rather unstable. Sometimes it does not converge at all or it might start to converge at a different point.

2.3 Questions and evaluation¶

Does the network look like it is overfitting or underfitting?

The network is severely overfitting. From the visualisation of training curves and reported final loss and accuracies, we can see that the network memorized training data perfectly, but accuracy for test data is has reached plateo at 0.705 (this number varies each training).

Is what we have here a good classifier? Could it be used in a realistic application?

We think that the classifier is pretty decent. For a problem with 10 classes, the probability of the correct choice of a class at random is 10% and the accuracy of our classifier is 70.5%.

It depends on the goal of a specific application of this classifier if it is good enough to be used in real

life. Since our task is recognition of spoken digits, which is not that hard for humans, the users of the system in which the classifier is going to be used is would probably expect a higher accuracy. One possible application is speech recognition. Then it is sensible to suggest that from the user perspective the system which makes errors in 3 of 10 cases is insufficient.

Do you think there is enough training data compared to the dimensions of the data and the number of parameters?

We have 2000 training samples with 6000 features and around 1.5 million parameters, which is \$O(100)\$ times larger than the number of samples. So, our problem is high-dimensional with the number of observations less than the number of features. The model with 1.5 million parameters seems to be rather complex. Since the network severely overfits, we think that this model's capacity is too large for the used data set. It can memorize training samples well, thus generalization suffers.

It is a good idea to try a simpler model for this task.

How could the classifier be improved?

As we started to argue in the previous question, it is sensible to try a simpler model - fewer filters and/or layers. Also different kernel sizes can be tried.

Another common solution to the overfitting problem is the addition of regularization. It can be L1 or L2 regularization or dropout. Also from the convergence curves, we see that for a very long time hardly any learning occurs, but then training converges quite rapidly. This might be a sign that our error landscape is challenging (for instance it might have plateaus and saddle points, local vs global structure problem). We use SDG optimizer, which is known to be affected by these issues, hence it is sensible to try another optimizer, e.g. Adam.

Batch normalization can speed up the convergence as well because it scales the input to belong to the "most non-linear" interval of activation functions input.

Also, different normalisation technique might significantly improve the convergence. We kept normalization to [-1,1] interval as was suggested initially, but also tried [0,1] (it didn't work at all) and zero mean + unit variance normalization of features (wavelengths). The latter worked noticeably better, however, we normalized the whole data set before splitting into training and test data. This is not the correct way to normalize - it should be done with training set statistics for both train and test sets to prevent the spread of test data information to the model during training. For proper normalization, we need to dive into Pytorch objects which would require additional time.

Lastly, more data is always better, so we could try to get a bigger dataset :)

The free spoken digits datasets has recordings from several different speakers. Is the test set accuracy a good measure of how well the trained network would perform for recognizing the voice of a new speaker? And if not, how could that be tested instead?

We think that the fact that no new speakers appear in our test set is indeed a flaw in our evaluation. We can split the data differently: leave the balanced distribution of digits in training and test sets, but do not mix the same speaker between these datasets. Then test accuracy would be more informative evaluation measure.

Also, we could try to look for data augmentation methods for speech recordings. If such methods exist, the addition of data augmentation should help to build a more accurate and robust model. And because of augmentation, we would have some "new" voices in the test set.

2.4 Variations¶

One way in which the training might be improved is with dropout or with batch normalization.

Make a copy of the network architecture below, and add dropout

Hint: see chapter 7.1 for an example that uses dropout. In [23]: num epochs = 200net dropout = torch.nn.Sequential(nn.Convld(1, 4, kernel size=5), nn.ReLU(), nn.AvgPool1d(kernel size=2, stride=2), nn.Conv1d(4, 8, kernel size=5), nn.ReLU(), nn.AvgPool1d(kernel size=2, stride=2), nn.Conv1d(8, 16, kernel size=5), nn.ReLU(), nn.AvgPool1d(kernel size=2, stride=2), nn.Conv1d(16, 32, kernel size=5), nn.ReLU(), nn.AvgPool1d(kernel_size=2, stride=2), nn.Flatten(), nn.Linear(11872, 128), nn.ReLU(), nn.Dropout(0.5), nn.Linear(128, 64), nn.ReLU(), nn.Dropout(0.5), nn.Linear(64, 10)) train(net dropout, train iter, test iter, num epochs, lr) loss 0.035, train acc 0.992, test acc 0.733

How does dropout change the results?

999.4 examples/sec on cpu

Addition of dropout slowed down the convergence and made it less steady. Also, train and test accuracy curves started to diverge later and for quite a few epochs test set accuracy was slightly higher. In the end, we got a 4.2% increase in test accuracy, which is rather good. We can also see that training accuracy hasn't reached 100%. Probably if we continued training for longer, we would be able to get slightly better accuracy.

We tried other dropout options as well. Decrease of FC layers dropout to 0.3 fails to suppress overfitting. With 0.4 the results were not very different, but the final accuracy seemed worse. We also tried adding a small dropout (0.1-0.2) to convolution layers, the network was not able to learn.

Make a copy of the original network architecture, and add batch normalization to all convolutional and linear layers.

```
nn.Convld(8, 16, kernel_size=5), nn.BatchNormld(16), nn.ReLU(),
    nn.AvgPoolld(kernel_size=2, stride=2),
    nn.Convld(16, 32, kernel_size=5), nn.BatchNormld(32), nn.ReLU(),
    nn.AvgPoolld(kernel_size=2, stride=2),
    nn.Flatten(),
    nn.Linear(11872, 128), nn.BatchNormld(128), nn.ReLU(),
    nn.Linear(128, 64), nn.BatchNormld(64), nn.ReLU(),
    nn.Linear(64, 10))
train(net_batchnorm, train_iter, test_iter, num_epochs, lr)

loss 0.003, train acc 1.000, test acc 0.773
801.1 examples/sec on cpu
```

How does batch normalization change the results?

Batch normalization increased convergence dramatically (already around 7th epoch) and significantly improved the final test accuracy. Hurray!

2.5 Bonus: feature extraction¶

Given enough training data a deep neural network can learn to extract features from raw data like audio and images. However, in some cases it is still necessary to do manual feature extraction. For speech recognition, a popular class of features are MFCCs.

Here is code to extract these features. You will need to install the $python_speech_features$ first.

```
In [25]:
```

```
pip install python_speech_features
```

Requirement already satisfied: python_speech_features in /home/denise/anaconda3/lib/python3.7/site-packages (0.6)

Note: you may need to restart the kernel to use updated packages.

```
In [26]:
```

```
from python_speech_features import mfcc

def load_waveform_mfcc(file, size = 6000):
    samplerate, waveform = wavfile.read(file)
    waveform = np.pad(waveform,(0,size))[0:size] / 32768
    return np.transpose(mfcc(waveform, samplerate))
```

Implement a variation of the dataset that uses these features

```
def init (self, data dir):
        digits x = []
        digits y = []
        for file in os.listdir(data dir):
            if file.endswith(".wav"):
                waveform = load waveform mfcc(os.path.join(data dir, file))
                label = int(file[0])
                digits x.append(waveform)
                digits y.append(label)
        # convert to torch tensors
        self.x = torch.from numpy(np.array(digits x, dtype=np.float32))
        self.x = self.x.unsqueeze(1) # One channel
        self.y = torch.from numpy(np.array(digits y))
    def len (self):
        return len(self.x)
    def getitem (self, idx):
        return self.x[idx], self.y[idx]
data mfcc = SpokenDigitsMFCC(data path)
train count mfcc = int(len(data_mfcc) * train_prop)
train mfcc, test mfcc = torch.utils.data.random split(data mfcc, [train count mfcc,
len(data mfcc)-train count mfcc])
train iter mfcc = torch.utils.data.DataLoader(train mfcc, **data params)
test iter mfcc = torch.utils.data.DataLoader(test mfcc, **data params)
The MFCC features will have 13 channels instead of 1 (the unsqueeze operation is not needed).
Inspect the shape of the data, and define a new network architecture that accepts data
In [28]:
```

with this shape

```
import matplotlib.pyplot as plt
# Since SpokenDigits returns the number of samples in __len__ method, we can just
use len(data) to
# obtain this number
print('The number of data samples is:',len(data mfcc))
# To check a sample dimensionality we use the size() method of the first sample
print('The dimension of a single sample is:',data mfcc.x[0].size())
# To obtain number of classes, we check the number of unique items in data.y array
print('The number of classes is:',len(np.unique(data_mfcc.y.numpy())))
data_mfcc.x[0][0]
plt.figure()
plt.plot(data.x[0][0])
plt.title("Original normalized data - first sample")
plt.show()
```

```
plt.figure()
plt.imshow(data mfcc.x[0][0])
plt.title("MFCC data with 13 channels - first sample")
plt.show()
The number of data samples is: 3000
The dimension of a single sample is: torch.Size([1, 13, 74])
The number of classes is: 10
In [29]:
# new architecture for new shapes
# adjusted copy of the batch normalization
class Reshape(torch.nn.Module):
    def forward(self, x):
        return x.view(-1,1,13,74)
num epochs = 20
net batchnorm mfcc = torch.nn.Sequential(
    Reshape(), # with reshape it is slightly more robust
    nn.Conv2d(1, 8, kernel_size=(2,3)), nn.BatchNorm2d(8), nn.ReLU(),
    nn.AvgPool2d(kernel size=2, stride=2),
    nn.Conv2d(8, 16, kernel size=3), nn.BatchNorm2d(16), nn.ReLU(),
    # no AvgPool here, because the channels shrink too fast.
    nn.Conv2d(16, 32, kernel size=3), nn.BatchNorm2d(32), nn.ReLU(),
    nn.AvgPool2d(kernel size=2, stride=2),
    nn.Flatten(),
    nn.Linear(512, 128), nn.ReLU(),
    nn.Linear(128, 64), nn.ReLU(),
    nn.Linear(64, 10))
X = torch.randn(size=(1, 1, 13, 74), dtype=torch.float32)
for layer in net batchnorm mfcc:
    X = layer(X)
    print(layer.__class__.__name__,'output shape: \t',X.shape)
Reshape output shape:
                         torch.Size([1, 1, 13, 74])
Conv2d output shape:
                         torch.Size([1, 8, 12, 72])
BatchNorm2d output shape:
                                 torch.Size([1, 8, 12, 72])
ReLU output shape:
                         torch.Size([1, 8, 12, 72])
AvgPool2d output shape:
                                 torch.Size([1, 8, 6, 36])
Conv2d output shape:
                         torch.Size([1, 16, 4, 34])
```

```
BatchNorm2d output shape: torch.Size([1, 16, 4, 34])
ReLU output shape: torch.Size([1, 16, 4, 34])
```

Conv2d output shape: torch.Size([1, 32, 2, 32])

BatchNorm2d output shape: torch.Size([1, 32, 2, 32])

ReLU output shape: torch.Size([1, 32, 2, 32])

AvgPool2d output shape: torch.Size([1, 32, 1, 16])

Flatten output shape: torch.Size([1, 512])
Linear output shape: torch.Size([1, 128])
ReLU output shape: torch.Size([1, 128])
Linear output shape: torch.Size([1, 64])
ReLU output shape: torch.Size([1, 64])
Linear output shape: torch.Size([1, 10])

Train the network with the mfcc features.

6491.4 examples/sec on cpu

```
In [30]:
    train(net_batchnorm_mfcc, train_iter_mfcc, test_iter_mfcc, num_epochs, lr)
    loss 0.003, train acc 1.000, test acc 0.972
```

Is there a neural-network based alternative to mfcc features?

Considering that (almost) all scientific papers on speech recognistion use mfcc or an equivalent transformation on their input data, we think that there exists no equivalent alternative based on neural-networks. In any case not a method that works better than converting your input data once form .wav format to mfcc format. You might be able to make it if you make custom layers to calculate the non-linear steps. Below you find the steps to calculate the mfcc transformation from the wikipedia page. In addition some conceptual ideas about how such a neural network could like.

- 1. Take the Fourier transform of (a windowed excerpt of) a signal.
 - For this step a neural network layer might just do the fourier transform, but the windowsize could be one of the parameters the neural network minimizes error for.
 Taking a window from the original data looks a lot like a 1d convolution.
- 2. Map the powers of the spectrum obtained above onto the mel scale, using triangular overlapping windows.
 - Triangular overlapping windows might be hard to spontaniously arise in the neural network, but you could hardcode something similar. This would be similar to a 2d convolutional layer, instead with a triangular shape instead of rectangular.
- 3. Take the logs of the powers at each of the mel frequencies.
 - taking a log function is not very linear, but I have no objection to making a layer that just takes the log. It might be equivalent to an activation function.
- 4. Take the discrete cosine transform of the list of mel log powers, as if it were a signal.
 - This would be similar to an activation function like the sigmoid, only then a cosine is used.
- 5. The MFCCs are the amplitudes of the resulting spectrum.
 - the output would be an image. (For example for the data in this assignment a 13 \times 74 shaped image)

The end¶

Well done! Please double check the instructions at the top before you submit your results.