

ML to predict time-to-failure of lab earthquakes from acoustic emissions

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Key topics and main questions



O

Data availability

What were the available data?

04

Features engineering

What were the key features for the learning?

02

Data handling

How did we handle raw data?

05

Model and Training

What model was used for prediction and how did we handle parameters tuning?

03

Data pre-processing

How did we build the dataset for training?

06

Performance and conclusions

How does the model perform? Holes in the dataset and cutting it.



Project overview

Why?

Traditional statistical methods do not seem to work in accurately predicting seismic events in a short-term window. Conversely, empirical evidence suggests that **Machine Learning can** be effectively used to **predict time-to-failure** in laboratory earthquakes.

Our motivating idea

2019: Los Alamos National Laboratory hosted a Kaggle competition.

<u>Objective</u>: predict the time remaining before laboratory earthquakes occur, based on real-time seismic acoustic data. This was a **significant open question** at the time, attracting interest from numerous research institutions.

Our goal: reproduce the state-of-the-art model at that time and then develop our own, learning from new data (La Sapienza Rock Mechanics Laboratory) to achieve a better performance.



Data

Data from La Sapienza Rock Mechanics Laboratory are acoustic emissions collected using the

TiePie oscilloscope.

6 channels:

Shear Stress, PU2, SL2, SR2, SR1, SL1.

Sample rate: 6 MHz

Float type: 32 bit

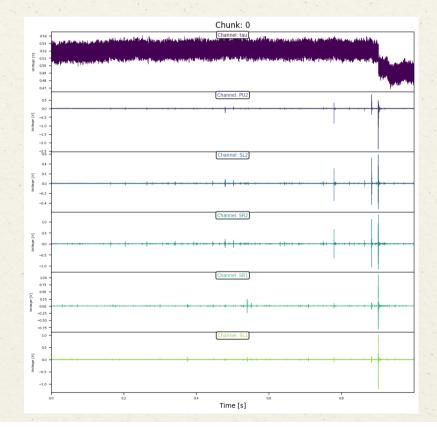
Organised in chunks

1 second each

200 chunks

→ 1.2 billion samples

Binary format





Data handling

Inspired by the Kaggle competition, we processed raw data to achieve a dataset in the format:

Acoustic emissions	Time to failure

For acoustic emissions we selected the channel (not the ST) with the highest variance. Due to memory constraints, we were unable to compute the variance directly over a tensor of 1.2

billion elements:



Data handling

Chosen channel: PU2

Converting from voltage [V] to arbitrary units (integer representation):

- $N = 2^{\text{# target bits}} = 2^{16} = 65536$
- Range of target arbitrary units: [0, 65536]

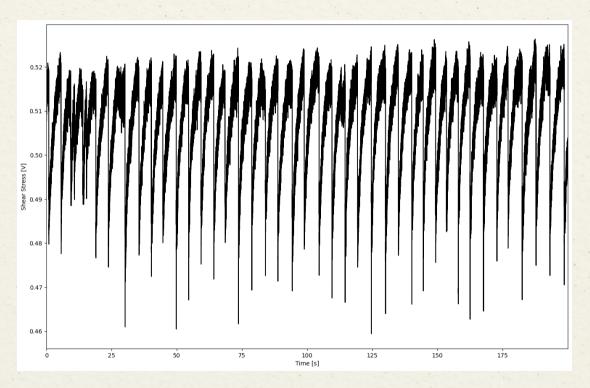
$$\mathrm{AU} = \left\lfloor \frac{(V - V_{\mathrm{min}})}{(V_{\mathrm{max}} - V_{\mathrm{min}})} \times (N - 1) \right\rfloor$$

Instead, to build the time-to-failure signal, we had to downsample the data recorded by the first channel due to memory constraints, (we were not able to perform the necessary operations on the full-resolution signal).

Interpolation.



Objective: to build the time-to-failure array Input: shear stress tensor of shape (1.2b,)





Downsampling: 6MHz → 1KHz

We now want to isolate the peaks to detect drops in shear stress.

Smoothing using a moving average



0.45

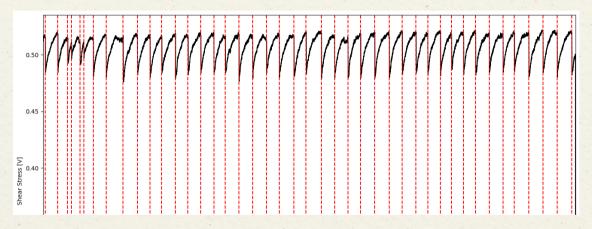
```
def moving_average(ST, window_size):
    # keep the original dimension while smoothing
    return np.convolve(ST, np.ones(window_size)/window_size, mode='same')

denoised_shear_stress = moving_average(shear_stress,50)
```



Identify Rapid Changes in Stress

(Through derivative over a sliding window)



After detecting picks in the shear stress, we can build the time-to-failure array. Remember: the shear stress is now sampled at 1 KHz.



Shape of time-to-failure array: (200k,)

We now want to upsample the time-to-failure array back to 6MHz.

<u>Why</u>: later, we will perform some analysis on model's performance using less data (including downsampling). For this reason, we want to start with all the data available.



Upsampling usually introduces some error, which would not be ideal. Just because we lack sufficient memory, we should not compromise testing the model's full potential.

Great news: we are lucky. We do not actually need to interpolate the data since we already know how the time-to-failure evolves over time (it is time-dependent, after all).

It's like expanding the vector through time, but we already know the values because time is continuous. The only thing we need to consider is that **we want 6,000 points for each point** in the downsampled array (since 6 MHz / 1 kHz = 6,000).



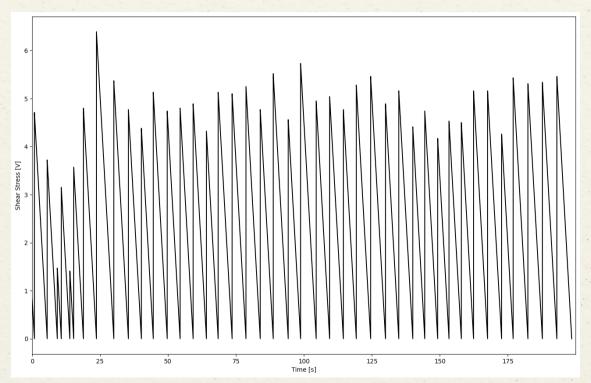
Here is how we did it:

```
# Compute the number of points to insert between each original point
points_each_point = int(fs / fs_red)
# Initialize the dense TTF array with a size scaled by the upsampling factor
TTF_dense = np.zeros(int(len(TTF) * points_each_point))
# Loop through the TTF array until the second—to—last element
for k in range(len(TTF) - 1):
   # Define the start and end indices in the dense array for the current segment
   start_index = k * points_each_point
   end_index = (k + 1) * points_each_point
   # If the current value is infinite, fill the remaining TTF_dense with infinity and exit the loop
   if np.isinf(TTF[k]):
       TTF dense[start index:] = np.inf
       break # No need to continue interpolation past this point
   # Compute the step size (delta) between two consecutive values in the original TTF array
   delta = (TTF[k + 1] - TTF[k]) / points_each_point
   # Linearly interpolate the values between TTF[k] and TTF[k+1]
   for i in range(points_each_point):
      TTF_dense[start_index + i] = TTF[k] + i * delta
```

Now our time-to-failure is (almost) ready.



Additionally, we remove the infinite values from the time-to-failure array. Result:



(We also ensure the corresponding last elements are removed from the acoustic data array too)



Features engineering

Input: time-to-failure and acoustic data arrays. Both of length 1,191,240,001.

We could calculate a vast number of features from the acoustic data. However, we selected 30 commonly used features and performed optimization to identify the best subset of features for learning, limiting the subset size to 5.

The results showed that the model achieved better performance when using the following set of sample features:

- Number of picks in the signal
- 20th percentile of the signal's standard deviation over a sliding window
- Mean (over time) Mel-Frequency Cepstral Coefficient 4 (out of 20)
- Mean (over time) Mel-Frequency Cepstral Coefficient 18 (out of 20)



Building dataset

- The full signal was divided into segments, to the tune of size 150,000 points (= 0.025 sec).
- For each segment, the selected features were calculated.
- The target value for each segment was defined as the time-to-failure at the end of the segment.
- Organised in Pandas dataframes.

TR	TRAINING DATAFRAME					
<cla< th=""><th colspan="6"><class 'pandas.core.frame.dataframe'=""></class></th></cla<>	<class 'pandas.core.frame.dataframe'=""></class>					
Rang	RangeIndex: 6316 entries, 0 to 6315					
Data	Data columns (total 6 columns):					
#	Column	Non-Null Count				
0	<pre>var_num_peaks_2_denoise_simple</pre>	6316 non-null				
1	<pre>var_percentile_roll50_std_20</pre>	6316 non-null				
2	<pre>var_mfcc_mean18</pre>	6316 non-null				
3	<pre>var_mfcc_mean4</pre>	6316 non-null				
4	start	6316 non-null				
5	target	6316 non-null				

TEST_DATAFRAME					
<pre><class 'pandas.core.frame.dataframe'=""></class></pre>					
RangeIndex: 1624 entries, 0 to 1623					
Data columns (total 6 columns):					
# Column	Non-Null Count				
<pre>0 var_num_peaks_2_denoise_simple</pre>	1624 non-null				
<pre>1 var_percentile_roll50_std_20</pre>	1624 non-null				
<pre>2 var_mfcc_mean18</pre>	1624 non-null				
5 target	1624 non-null				
dtypes: float64(6)					



Training

- Gradient Boosting
- 3-fold cross-validation (→ validation metrics)
- Training size: Approximately 80% of the full dataset (ensuring cuts align with full events)
- Hyperparameter optimization: Selecting the best parameters from a large search space
- Early stopping

```
from lightgbm import LGBMRegressor
from sklearn.model_selection import GridSearchCV, KFold

param_grid = {
    'num_leaves': [4, 8, 16],
    'min_data_in_leaf': [5, 10, 20],
    'learning_rate': [0.01, 0.02, 0.05],
    'feature_fraction': [0.7, 0.9],
    'bagging_fraction': [0.5, 0.7],
    'reg_alpha': [0, 0.1, 0.5],
    'reg_lambda': [0, 0.1, 0.5]
}
```

```
base_model = LGBMRegressor(
    boosting_type='gbdt',
    objective='fair',
    max_depth=-1,
    boosting='gbdt',
    bagging_freq=1,
    bagging_seed=0,
    metric='mae',
    verbosity=-1,
    max_bin=500,
    seed=0,
    n_jobs=-1
)
```



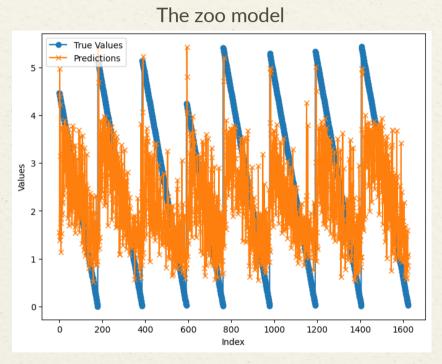
Performance

Our model vs. state-of-the-art model at the time of the Kaggle competition:

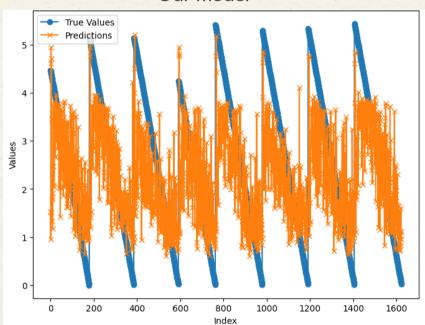
Metric on test set	B. Rouet-Leduc model	The Zoo model	Our model
MAE	0.9627	0.9558	0.9286
MSE	1.2304	1.2188	1.1930
R ²	0.3208	0.3318	0.3614



Performance



Our model





One more thing...



Holes in the dataset

Is the Sampling Rate Too High? Selecting x samples every 40:

Metric on test set	x = 40	x = 30	x = 20	x = 10	x = 1
MAE	0.9286	0.9263	0.9369	0.9548	1.1061
MSE	1.1930	1.1930	1.2049	1.2226	1.3898
R ²	0.3614	0.3614	0.3487	0.3294	0.1334



Using less events

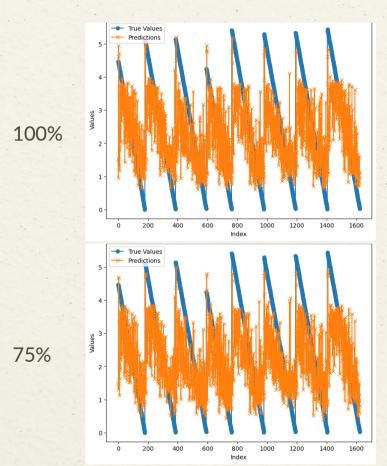
Using only fractions of the dataset:

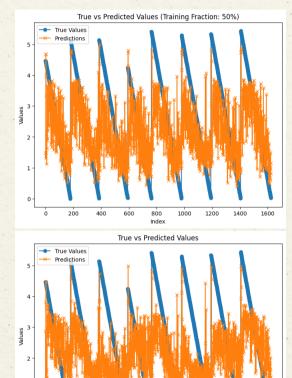
Metric on test	100%	75%	50%	25%	2.5%
MAE	0.9286	0.9395	0.9723	1.0083	1.1503
MSE	1.1930	1.2103	1.2410	1.2837	1.4220
R ²	0.3614	0.3428	0.3090	0.2607	0.0928

Holes in the dataset is better than reducing the dataset size by cutting it.

Using less events







50%

2.5%

