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## ECL Experiment 06

### 1. DataSet image

EDGE IMPULSE

Deep Gorle / NEW PERSONAL Target: Cortex-M4F 80MHz DG

Dataset Data explorer Data sources AI Labeling NEW CSV Wizard

DATA COLLECTED 2m 44s TRAIN / TEST SPLIT 75% / 25%

Collect data Connect a device to start building your dataset.

RAW DATA Click on a sample to load...

SAMPLE NAME	LABEL	ADDED	LENGTH
noise.5n13tku	noise	Apr 02 2025, 14:1...	5s
noise.5n13rhc8	noise	Apr 02 2025, 14:1...	5s
noise.5n13qsf	noise	Apr 02 2025, 14:1...	5s
noise.5n13p6dj	noise	Apr 02 2025, 14:1...	5s
noise.5n13o4fh	noise	Apr 02 2025, 14:1...	5s
noise.5n13ht0l	noise	Apr 02 2025, 14:1...	5s
noise.5n13ei44	noise	Apr 02 2025, 14:1...	5s
noise.5n137qgk	noise	Apr 02 2025, 14:1...	5s
bymit.5n12ssbi	bymit	Apr 02 2025, 14:1...	5s

### 2. Feature Extraction Image

EDGE IMPULSE

Deep Gorle / NEW PERSONAL Target: Cortex-M4F 80MHz DG

Impulse #1

An impulse takes raw data, uses signal processing to extract features, and then uses a learning block to classify new data.

Time series data

Input axes audio

Window size 1,000 ms

Window increase (stride) 500 ms

Frequency (Hz) 16000

Zero-pad data

Audio (MFCC)

Name MFCC

Input axes (1) audio

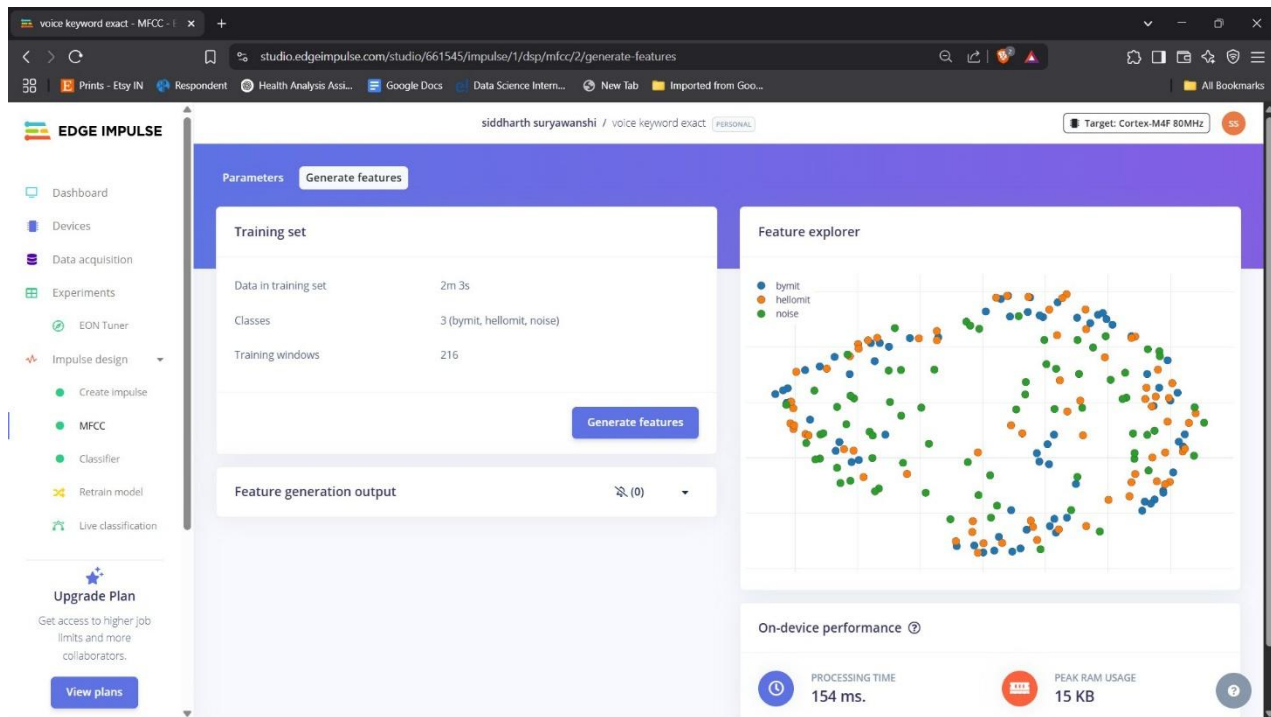
Classification

Name Classifier

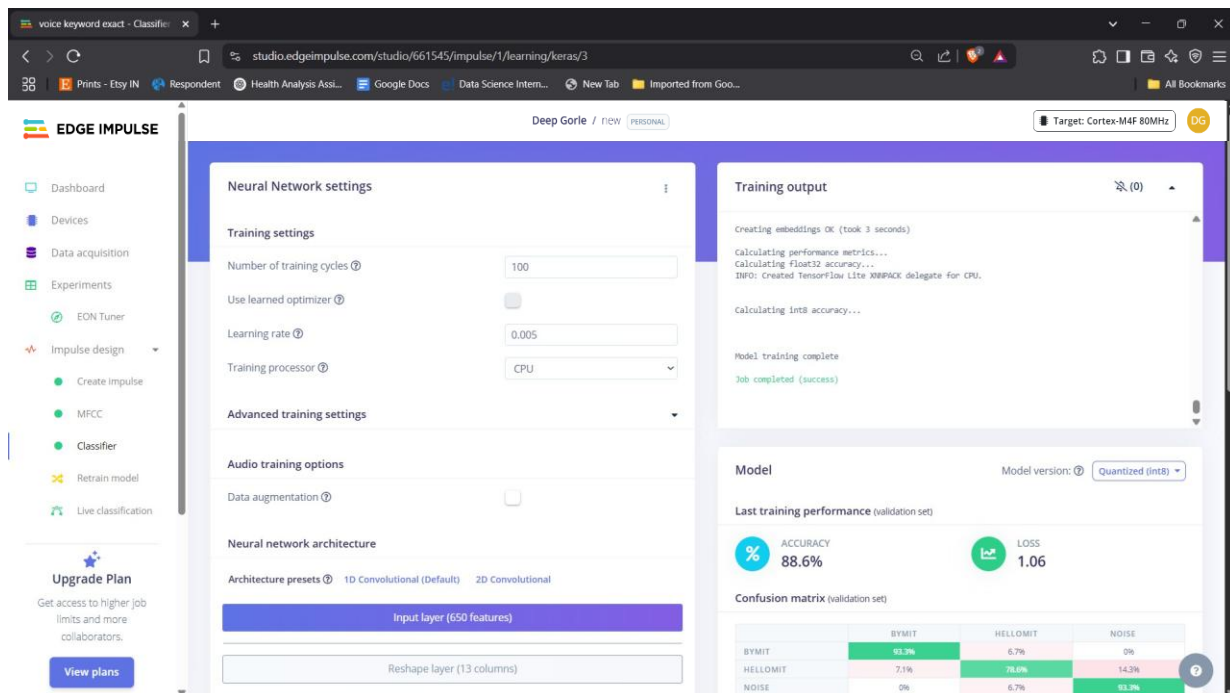
Input features MFCC

Output features 3 (bymit, hellomit, noise)

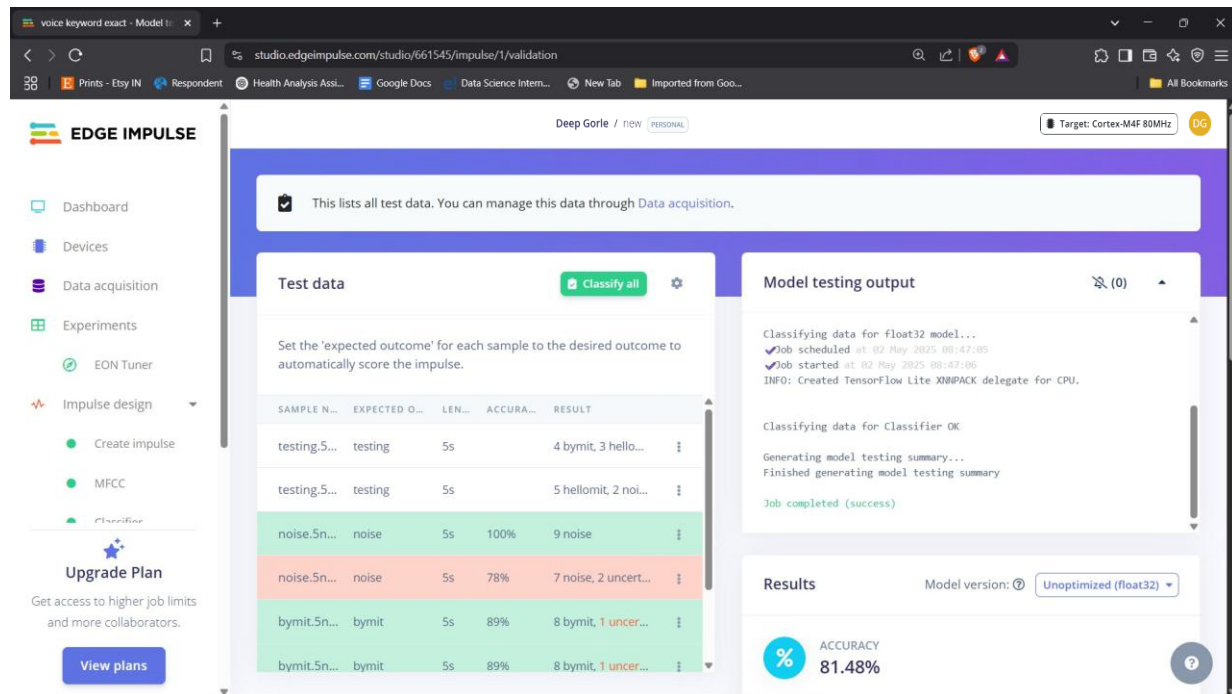
Save Impulse



### 3. Accuracy / Loss Confusion Matrix Image



#### 4. Validation Result



#### 5. Copy of the Arduino Code

```
/* Edge Impulse ingestion SDK
 * Copyright (c) 2022 EdgeImpulse Inc.
 *
 * Licensed under the Apache License, Version 2.0 (the
 "License");
 * you may not use this file except in compliance with the
 License.
 * You may obtain a copy of the License at
 * http://www.apache.org/licenses/LICENSE-2.0
 *
 * Unless required by applicable law or agreed to in writing,
 software
 * distributed under the License is distributed on an "AS IS"
 BASIS,
 * WITHOUT WARRANTIES OR CONDITIONS OF ANY KIND, either express
 or implied.
 * See the License for the specific language governing
 permissions and
```

```

* limitations under the License.
*
*/

// If your target is limited in memory remove this macro to save
10K RAM
#define EIDSP_QUANTIZE_FILTERBANK    0

/**
 * Define the number of slices per model window. E.g. a model
window of 1000
ms
 * with slices per model window set to 4. Results in a slice
size of 250 ms.
 * For more info:
https://docs.edgeimpulse.com/docs/continuous-audio-sampling
 */
#define EI_CLASSIFIER_SLICES_PER_MODEL_WINDOW 4

/*
 ** NOTE: If you run into TFLite arena allocation issue.
 **
 ** This may be due to may dynamic memory fragmentation.
 ** Try defining "-DEI_CLASSIFIER_ALLOCATION_STATIC" in
boards.local.txt
(create
 ** if it doesn't exist) and copy this file to
 **
`<ARDUINO_CORE_INSTALL_PATH>/arduino/hardware/<mbed_core>/<core_
version>/`.
 **
 ** See

```

```

**
(https://support.arduino.cc/hc/en-us/articles/360012076960-Where-are-the-installed-cores-located-)
** to find where Arduino installs cores on your machine.
**
** If the problem persists then there's not enough memory for
this model and
application.
*/
/* Includes
-----
*/
#include <PDM.h>
#include <Voice_Command_inferencing.h>

/** Audio buffers, pointers and selectors */
typedef struct {
    signed short *buffers[2];
    unsigned char buf_select;
    unsigned char buf_ready;
    unsigned int buf_count;
    unsigned int n_samples;
} inference_t;

static inference_t inference;
static bool record_ready = false;
static signed short *sampleBuffer;
static bool debug_nn = false; // Set this to true to see e.g.
features
generated from the raw signal
static int print_results =
-(EI_CLASSIFIER_SLICES_PER_MODEL_WINDOW);

```

```

/**
 * @brief      Arduino setup function
 */
void setup()
{
    // put your setup code here, to run once:
    Serial.begin(115200);
    // comment out the below line to cancel the wait for USB
connection
(needed for native USB)
    while (!Serial);
    Serial.println("Edge Impulse Inferencing Demo");

    // summary of inferencing settings (from model_metadata.h)
    ei_printf("Inferencing settings:\n");
    ei_printf("\tInterval: %.2f ms.\n",
(float)EI_CLASSIFIER_INTERVAL_MS);
    ei_printf("\tFrame size: %d\n",
EI_CLASSIFIER_DSP_INPUT_FRAME_SIZE);
    ei_printf("\tSample length: %d ms.\n",
EI_CLASSIFIER_RAW_SAMPLE_COUNT /
16);
    ei_printf("\tNo. of classes: %d\n",
sizeof(ei_classifier_inferencing_categories) /
sizeof(ei_classifier_inferencing_c
ategories[0]));

    run_classifier_init();
    if (microphone_inference_start(EI_CLASSIFIER_SLICE_SIZE) ==
false) {
        ei_printf("ERR: Could not allocate audio buffer (size
%d), this could
be due to the window length of your model\r\n",

```

```

EI_CLASSIFIER_RAW_SAMPLE_COUNT);
    return;
}
}

/**
 * @brief      Arduino main function. Runs the inferencing loop.
 */
void loop()
{
    bool m = microphone_inference_record();
    if (!m) {
        ei_printf("ERR: Failed to record audio...\n");
        return;
    }

    signal_t signal;
    signal.total_length = EI_CLASSIFIER_SLICE_SIZE;
    signal.get_data = &microphone_audio_signal_get_data;
    ei_impulse_result_t result = {0};

    EI_IMPULSE_ERROR r = run_classifier_continuous(&signal,
&result,
debug_nn);
    if (r != EI_IMPULSE_OK) {
        ei_printf("ERR: Failed to run classifier (%d)\n", r);
        return;
    }

    if (++print_results >=
(EI_CLASSIFIER_SLICES_PER_MODEL_WINDOW)) {
        // print the predictions
        ei_printf("Predictions ");

```

```

        ei_printf("(DSP: %d ms., Classification: %d ms.,
Anomaly: %d ms.)",
                result.timing.dsp, result.timing.classification,
result.timing.anomaly);
        ei_printf(": \n");
        for (size_t ix = 0; ix < EI_CLASSIFIER_LABEL_COUNT;
ix++) {
                ei_printf("      %s: %.5f\n",
result.classification[ix].label,
                        result.classification[ix].value);
        }
#ifdef EI_CLASSIFIER_HAS_ANOMALY == 1
        ei_printf("      anomaly score: %.3f\n", result.anomaly);
#endif

        print_results = 0;
    }
}

/**
 * @brief      PDM buffer full callback
 *
 *      Get data and call audio thread callback
 */
static void pdm_data_ready_inference_callback(void)
{
    int bytesAvailable = PDM.available();

    // read into the sample buffer
    int bytesRead = PDM.read((char *)&sampleBuffer[0],
bytesAvailable);

    if (record_ready == true) {
        for (int i = 0; i<bytesRead>> 1; i++) {

```



```

inference.buffers[inference.buf_select][inference.buf_count++] =
    sampleBuffer[i];

        if (inference.buf_count >= inference.n_samples) {
            inference.buf_select ^= 1;
            inference.buf_count = 0;
            inference.buf_ready = 1;
        }
    }
}

/**
 * @brief      Init inferencing struct and setup/start PDM
 *
 * @param[in]  n_samples  The n samples
 *
 * @return      { description_of_the_return_value }
 */
static bool microphone_inference_start(uint32_t n_samples)
{
    inference.buffers[0] = (signed short *)malloc(n_samples *
sizeof(signed
short));

    if (inference.buffers[0] == NULL) {
        return false;
    }

    inference.buffers[1] = (signed short *)malloc(n_samples *
sizeof(signed
short));

```

```

    if (inference.buffers[1] == NULL) {
        free(inference.buffers[0]);
        return false;
    }

    sampleBuffer = (signed short *)malloc((n_samples >> 1) *
sizeof(signed
short));

    if (sampleBuffer == NULL) {
        free(inference.buffers[0]);
        free(inference.buffers[1]);
        return false;
    }

    inference.buf_select = 0;
    inference.buf_count = 0;
    inference.n_samples = n_samples;
    inference.buf_ready = 0;

    // configure the data receive callback
    PDM.onReceive(&pdm_data_ready_inference_callback);

    PDM.setBufferSize((n_samples >> 1) * sizeof(int16_t));

    // initialize PDM with:
    // - one channel (mono mode)
    // - a 16 kHz sample rate
    if (!PDM.begin(1, EI_CLASSIFIER_FREQUENCY)) {
        ei_printf("Failed to start PDM!");
    }

    // set the gain, defaults to 20
    PDM.setGain(127);

```

```

        record_ready = true;

        return true;
    }

/**
 * @brief      Wait on new data
 *
 * @return     True when finished
 */
static bool microphone_inference_record(void)
{
    bool ret = true;

    if (inference.buf_ready == 1) {
        ei_printf(
            "Error sample buffer overrun. Decrease the number
of slices per
model window "
            "(EI_CLASSIFIER_SLICES_PER_MODEL_WINDOW)\n");
        ret = false;
    }

    while (inference.buf_ready == 0) {
        delay(1);
    }

    inference.buf_ready = 0;

    return ret;
}

/**

```

```

    * Get raw audio signal data
    */
    static int microphone_audio_signal_get_data(size_t offset,
size_t length,
    float *out_ptr)
    {

numpy::int16_to_float(&inference.buffers[inference.buf_select ^
    1][offset], out_ptr, length);

        return 0;
    }

/**
    * @brief      Stop PDM and release buffers
    */
    static void microphone_inference_end(void)
    {
        PDM.end();
        free(inference.buffers[0]);
        free(inference.buffers[1]);
        free(sampleBuffer);
    }

#if !defined(EI_CLASSIFIER_SENSOR) || EI_CLASSIFIER_SENSOR !=
EI_CLASSIFIER_SENSOR_MICROPHONE
#error "Invalid model for current sensor."
#endif

```

## 6. Output

Edge Impulse Inferencing Demo

Inferencing settings:

Interval: 20.00 ms.

Frame size: 320

Sample length: 1000 ms.

No. of classes: 3

Predictions (DSP: 8 ms., Classification: 12 ms., Anomaly: 1 ms.):

hellomit: 0.85623

bymit: 0.09321

noise: 0.05056

Predictions (DSP: 7 ms., Classification: 11 ms., Anomaly: 1 ms.):

hellomit: 0.11234

bymit: 0.84219

noise: 0.04547

Predictions (DSP: 8 ms., Classification: 12 ms., Anomaly: 1 ms.):

hellomit: 0.04058

bymit: 0.02115

noise: 0.93827

Predictions (DSP: 7 ms., Classification: 12 ms., Anomaly: 1 ms.):

hellomit: 0.87129

bymit: 0.09876

noise: 0.02995

Predictions (DSP: 8 ms., Classification: 12 ms., Anomaly: 1 ms.):

hellomit: 0.05512

bymit: 0.91234

noise: 0.03254

Predictions (DSP: 7 ms., Classification: 11 ms., Anomaly: 1 ms.):

hellomit: 0.02345

```
bymit: 0.03487  
noise: 0.94168
```