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Title

Keyword Spotting Project like "OK, Google," "Alexa," on Edge Devices using Microphone

Objective: Build a project to detect the keywords using a built-in sensor on Nano BLE Sense /
Mobile Phone

Tasks:

- Generate the dataset for keyword
- Configure BLE Sense / Mobile for Edge Impulse
- Building and Training a Model

Run the project Keyword Spotting like "OK, Google," "Alexa

Introduction

Edge Impulse is a development platform for machine learning on edge devices, targeted at developers who want to create intelligent device solutions. The "Hello World" equivalent in Edge Impulse would typically involve creating a simple machine learning model that can run on an edge device, like classifying sensor data or recognizing a basic pattern.

Materials Required

- Nano BLE Sense Board

ECL Experiment 6

The screenshot displays the Edge Impulse Studio web interface in a browser window. The address bar shows the URL: `studio.edgeimpulse.com/studio/681665/acquisition/training?page=1`. The interface is for a project named 'Sudip04' (Sudip04-project-1) with a target of 'Cortex-M4F 80MHz'.

Left Sidebar:

- Dashboard
- Devices
- Data acquisition
- Experiments
- EON Tuner
- Impulse design
- Upgrade Plan (Get access to higher job limits and more collaborators. View plans)

Main Content Area:

- Dataset:** DATA COLLEC... 2m 34s
- TRAIN / TES...:** 70% ...
- Collect data:** Connect a device to start building your dataset.
- Dataset Table:**

SAMPLE NAME	LABEL	ADDED	LENGTH
Noise.5pt6...	Noise	Apr 29 202...	5s
Noise.5pt6...	Noise	Apr 29 202...	5s
Noise.5pt6...	Noise	Apr 29 202...	5s
Noise.5pt6...	Noise	Apr 29 202...	5s

RAW DATA: Click on a sample to load...

Bottom Bar: Resume tutorial

Edge Impulse Studio interface for creating an impulse. The browser address bar shows `studio.edgeimpulse.com/studio/681665/impulse/1/create-impulse`. The target device is set to **Target: Cortex-M4F 80MHz**.

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):

Audio (MFCC)

- Name: MFCC
- Input axes (1): audio
- Signal: audio

Classification

- Name: Classifier
- Input features: ☒ MFCC
- Output features: 3 (No, Noise, OK Google)

Output features

- 3 (No, Noise, OK Google)

[Save impulse](#)

[Resume tutorial](#)

[View plans](#)

[Upgrade Plan](#)

Get access to higher job limits and more collaborators.

Microsoft Rewards

21:21 02-05-2025

Edge Impulse Studio interface for training a classifier. The browser address bar shows `studio.edgeimpulse.com/studio/681665/impulse/1/learning/keras/3`. The target device is set to **Target: Cortex-M4F 80MHz**.

Neural Network settings

Training settings

- Number of training cycles: 100
- Use learned optimizer: ☐
- Learning rate: 0.005
- Training processor: CPU

Advanced training settings

Audio training options

- Data augmentation: ☐

Training output

Model version: Quantized (int8)

Last training performance (validation set)

- ACCURACY: 97.4%
- LOSS: 0.07

Confusion matrix (validation set)

	NO	NOISE	OK GOOGLE
NO	91.7%	0%	8.3%
NOISE	0%	100%	0%
OK GOOGLE	0%	0%	100%
F1 SCORE	0.96	1.00	

[Resume tutorial](#)

Microsoft Rewards

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```

/* 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) {

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        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; 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.buffer[0]);
        free(inference.buffer[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