Code:

```
#define EIDSP_QUANTIZE_FILTERBANK   0
 */
#define EI_CLASSIFIER_SLICES_PER_MODEL_WINDOW 4



/* Includes
---------------------------------------------------------------- */
#include <PDM.h>
#include <TinyML_Arduino_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_categories[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);
        }
#if 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
```

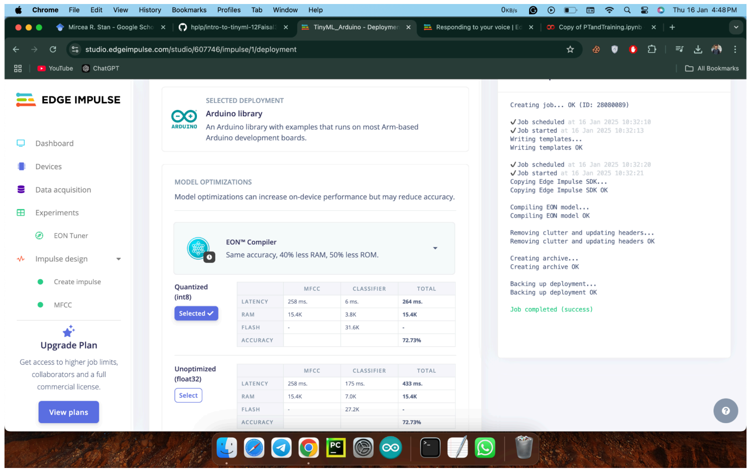
```
Output:

Predictions (DSP: 71 ms., Classification: 6 ms., Anomaly: 0 ms.):

   helloworld: 0.81250

   noise: 0.18750

Predictions (DSP: 71 ms., Classification: 6 ms., Anomaly: 0 ms.):

   helloworld: 0.92969

   noise: 0.07031

Predictions (DSP: 71 ms., Classification: 6 ms., Anomaly: 0 ms.):

   helloworld: 0.88281

   noise: 0.11719
```

```
Predictions (DSP: 71 ms., Classification: 6 ms., Anomaly: 0 ms.):

   helloworld: 0.87891

   noise: 0.12109

Predictions (DSP: 71 ms., Classification: 6 ms., Anomaly: 0 ms.):

   helloworld: 0.64453
```

Questions:
Does the model perform as accurately as expected on your smartphone? List a few methods to improve the model's accuracy.
Ans: no, to improve the accuracy increase the number of test inputs.

When building a model for resource-limited hardware, how do you balance fast inference times with acceptable model accuracy? What trade-offs did you encounter?
Ans: Methods to Improve Model Accuracy Some methods to improve the model's accuracy are:
Data Augmentation: Introduce variations in the dataset to make the model more robust.
Model Architecture: Use architectures like EfficientNet or MobileNet designed for optimal accuracy with lower computational cost.
Hyperparameter Tuning: Experiment with learning rates, batch sizes, and optimizers.
Transfer Learning: Use pre-trained models and fine-tune them for your task.
Regularization Techniques: Apply dropout, weight decay, or batch normalization to prevent overfitting.

Include screenshots of the training performance from step 6 of the deployment process.



Reflections: Share your experience deploying the model to your smartphone and Arduino board. Mention any technical difficulties or interesting observations.

Ans: less number of datasets and two classes (helloworld and noise)