

# **Audio Engine**

# **User Manual**

STM32G474 DSP Audio Playback System  
Version 2.0

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# Audio Engine User Manual

## Overview

The Audio Engine is a reusable, embedded DSP audio playback system designed for STM32 microcontrollers with I2S support (including the STM32G4, STM32F4, STM32H7 series and others) with audio output to digital amplifiers such as the MAX98357A.

### Key Features

- **Dual Format Support:** 8-bit and 16-bit audio playback
- **Flexible Modes:** Mono and stereo playback
- **DSP Filter Chain:** Runtime-configurable filters with fixed-point arithmetic
- **No FPU Required:** All DSP operations use integer math for MCU efficiency
- **Sample Rate:** Default 22 kHz (configurable)
- **DMA-Driven:** Efficient I2S streaming with double-buffering
- **Low Latency:** ~93 ms playback latency with 2048-sample buffer

### Core Capabilities

Feature	Specification
Sample Rates	22 kHz (default), configurable
Audio Depths	8-bit unsigned, 16-bit signed
Channels	Mono, Stereo
Buffer Size	2048 samples (ping-pong DMA)
Nyquist Frequency	11 kHz @ 22 kHz sample rate
Volume Control	Software configurable (0-3x gain)
Fade Effects	In/Out (~93 ms at 22 kHz)

---

## Quick Start

### 1. Initialize the Audio Engine

You **must** call `AudioEngine_Init()` to set up the audio engine with the required hardware callbacks. This function initializes all filter state and validates that the necessary hardware interface functions are provided:

```
#include "audio_engine.h"

// Step 1: I2S2 must be initialized via CubeMX
// (This is done automatically in MX_I2S2_Init())

// Step 2: Initialize the audio engine with hardware callbacks
```

```
PB_StatusTypeDef status = AudioEngine_Init(
    DAC_MasterSwitch,    // Function to control amplifier on/off
    ReadVolume,          // Function to read volume setting
    MX_I2S2_Init         // Function to re-initialize I2S if needed
);

if( status != PB_Idle ) {
    // Handle initialization error
}

// Step 3: Configure filters (optional, defaults are pre-set)
SetLpf16BitLevel(LPF_Soft);           // Set filter aggressiveness
SetFilterConfig(&my_filter_config);   // Apply complete config

// Audio engine is now ready to play samples
```

## 2. Play a 16-bit Audio Sample

```
// Assuming 'doorbell_sound' is a 16-bit mono WAV sample in flash memory
// 44,100 samples = ~2 seconds @ 22 kHz, 16-bit mono

extern const uint8_t doorbell_sound[];
extern const uint32_t doorbell_sound_size;

PB_StatusTypeDef result = PlaySample(
    doorbell_sound,        // Pointer to audio data
    doorbell_sound_size,   // Total samples (all channels combined)
    22000,                // Sample rate (Hz)
    16,                   // Bit depth (16 = 16-bit)
    Mode_stereo            // Stereo playback
);

// Wait for playback to complete
WaitForSampleEnd();
```

## 3. Configure Filters

```
// Get current filter configuration
FilterConfig_TypeDef cfg;
GetFilterConfig(&cfg);

// Adjust filter levels
cfg.enable_16bit_biquad_lpf = 1;    // Enable 16-bit LPF
cfg.enable_soft_clipping = 1;       // Enable soft clipping to prevent distortion
cfg.lpf_16bit_level = LPF_Medium;   // Medium filtering strength

// Apply new configuration
SetFilterConfig(&cfg);

// Or use convenience function for LPF level
SetLpf16BitLevel(LPF_Aggressive);   // Stronger filtering
```

---

## Architecture

### System Block Diagram

The audio playback system follows a clear data flow from flash memory through DSP processing to the I2S output. The DMA operates in ping-pong mode, processing audio chunks as they're transmitted, ensuring continuous playback without CPU blocking.

*[svglib not installed - install with: pip install svglib]*

## Filter Chain Stages (16-bit Audio)

### 1. Biquad Low-Pass Filter *(Optional - enable\_16bit\_biquad\_lpf)*

- Second-order IIR filter
- Runtime-configurable aggressiveness: Very Soft → Aggressive
- Warm-up: 16 passes of first sample to prevent startup artefacts
- Cutoff range (22 kHz fs, approx): ~2.6 kHz (Very Soft), ~1.4 kHz (Soft), ~0.9 kHz (Medium), ~0.2 kHz (Aggressive)
- 64-bit accumulator in the biquad path to prevent overflow with aggressive settings

### 2. DC Blocking Filter *(Selectable - enable\_soft\_dc\_filter\_16bit)*

- Removes DC offset and very low frequencies
- Two variants: standard (44 Hz) or soft (22 Hz)
- Prevents output drift
- Always active in one of the two modes

### 3. Air Effect (High-Shelf) *(Optional - enable\_air\_effect)*

- Adds presence and brightness to audio
- Runtime-adjustable boost (+1 dB, +2 dB, +3 dB presets)
- Disabled by default

### 4. Fade In/Out *(Always Active)*

- Quadratic power curve ramp
- Default: 2048 samples (~93 ms @ 22 kHz)
- Smooth entry/exit for audio transitions

### 5. Noise Gate *(Optional - enable\_noise\_gate)*

- Mutes samples below  $\pm 512$  amplitude
- Suppresses quantization noise during silence
- Disabled by default

### 6. Soft Clipping *(Optional - enable\_soft\_clipping)*

- Smooth cubic curve limiting above  $\pm 28,000$
- Prevents harsh digital clipping
- Musical, natural-sounding compression
- Recommended: keep enabled

### 7. Volume Scaling *(Always Active)*

- Integer multiplication (0–3x gain)
- Read from hardware GPIO (3-level selector)
- Applied per-sample

## Filter Chain Stages (8-bit Audio)

### 1. 8-bit to 16-bit Conversion with Dithering *(Always Active)*

- TPDF (Triangular PDF) dithering reduces quantization noise
  - Upsamples to internal 16-bit working format
2. **One-Pole Low-Pass Filter** (*Optional - enable\_8bit\_lpf*)
- One-pole IIR (alpha range: 0.625 to 0.9375)
  - Separate aggressiveness levels for 8-bit audio
  - Custom alpha supported via `SetLpf8BitCustomAlpha()` or `CalcLpf8BitAlphaFromCutoff()`
3. **Makeup Gain** (*Always Active when LPF enabled*)
- Post-LPF amplitude compensation (~1.08x default)
  - Configurable via `SetLpfMakeupGain8Bit()`
4. **DC Blocking & Remaining Stages**
- Same as 16-bit path (steps 2–7)
  - Air Effect, Fade, Noise Gate, Soft Clipping, Volume Scaling
- 

## API Reference

### Enumeration Types

#### ``PB_StatusTypeDef``

Audio playback state enumeration.

```
typedef enum {
    PB_Idle,           // No audio playing
    PB_Error,          // Error during playback
    PB_Playing,        // Audio actively playing
    PB_Paused,         // Playback paused
    PB_PlayingFailed   // Playback failed to start
} PB_StatusTypeDef;
```

#### ``PB_ModeTypeDef``

Playback channel mode.

```
typedef enum {
    Mode_stereo,       // Stereo (2-channel) playback
    Mode_mono          // Mono (single-channel) playback
} PB_ModeTypeDef;
```

#### ``LPF_Level``

Low-pass filter aggressiveness level for 16-bit and 8-bit LPFs.

```
typedef enum {
    LPF_Off,           // Filtering disabled
    LPF_VerySoft,      // Minimal filtering ( $\alpha = 0.625$ )
    LPF_Soft,          // Gentle filtering ( $\alpha \approx 0.80$ )
    LPF_Medium,        // Balanced filtering ( $\alpha = 0.875$ )
    LPF_Firm,          // Firm filtering ( $\alpha \approx 0.92$ )
}
```

```

    LPF_Aggressive,    // Strong filtering ( $\alpha \approx 0.97$ )
    LPF_Custom         // Use custom alpha (SetLpf16BitCustomAlpha or SetLpf8BitCustomAlpha)
} LPF_Level;

```

## Structure Types

### `FilterConfig\_TypeDef`

Runtime filter configuration structure.

```

typedef struct {
    uint8_t enable_16bit_biquad_lpf; // Enable/disable 16-bit biquad LPF
    uint8_t enable_soft_dc_filter_16bit; // Use softer DC blocking (22 Hz vs 44 Hz)
    uint8_t enable_8bit_lpf; // Enable/disable 8-bit one-pole LPF
    uint8_t enable_noise_gate; // Enable/disable noise gate
    uint8_t enable_soft_clipping; // Enable/disable soft clipping
    uint8_t enable_air_effect; // Enable/disable air effect
    uint32_t lpf_makeup_gain_q16; // Post-LPF gain in Q16 fixed-point
    LPF_Level lpf_16bit_level; // 16-bit LPF aggressiveness
    uint16_t lpf_16bit_custom_alpha; // Custom alpha for 16-bit LPF
    LPF_Level lpf_8bit_level; // 8-bit LPF aggressiveness
    uint16_t lpf_8bit_custom_alpha; // Custom alpha for 8-bit LPF
} FilterConfig_TypeDef;

```

### Field Descriptions:

- `lpf_makeup_gain_q16` : Gain value in Q16 format (65536 = 1.0x). Default: 70779 (~1.08x)
- `lpf_16bit_level` : Filter aggressiveness affects cutoff frequency and stop-band attenuation
- `lpf_16bit_custom_alpha` : Custom alpha used when `lpf_16bit_level = LPF_Custom`
- `lpf_8bit_level` : Filter aggressiveness for the 8-bit one-pole LPF
- `lpf_8bit_custom_alpha` : Custom alpha used when `lpf_8bit_level = LPF_Custom`

### `AudioEngine\_HandleTypeDef`

Audio engine state handle (for initialization).

```

typedef struct {
    I2S_HandleTypeDef *hi2s; // Pointer to I2S HAL handle
    int16_t *pb_buffer; // Playback buffer (2048 samples)
    uint32_t playback_speed; // Default playback speed (Hz)
} AudioEngine_HandleTypeDef;

```

## Function Reference

### Hardware Setup (Done in CubeMX + main.c)

Before playing audio, ensure:

1. **I2S2 is configured** in CubeMX (22 kHz, 16-bit, DMA enabled)
2. `AudioEngine_Init()` **is called** with function pointers for:
  - DAC on/off control
  - Volume reading
  - I2S re-initialization

3. **Filters are configured** to desired settings (optional; defaults are applied by `AudioEngine_Init()` )

Calling `AudioEngine_Init()` is **required** before any audio playback. It initializes the filter state, validates hardware callbacks, and sets up default filter configuration.

## Playback Control

### `AudioEngine_Init()`

Initialize the audio engine with required hardware callbacks.

```
PB_StatusTypeDef AudioEngine_Init(
    DAC_SwitchFunc dac_switch,
    ReadVolumeFunc read_volume,
    I2S_InitFunc i2s_init
);
```

#### Parameters:

- `dac_switch` : Function pointer for controlling amplifier on/off (GPIO control)
- `read_volume` : Function pointer for reading current volume level
- `i2s_init` : Function pointer for I2S peripheral re-initialization

#### Returns:

- `PB_Idle` if initialization successful
- `PB_Error` if any function pointer is NULL

#### Important Notes:

- **Must be called once** before any call to `PlaySample()` or other playback functions
- Initializes all filter state variables and resets playback status
- Sets up default filter configuration (can be overridden with `SetFilterConfig()` )
- Validates that all required hardware callbacks are provided
- Does not start audio playback itself

#### Example:

```
#include "audio_engine.h"

// Define these functions in your application
void DAC_MasterSwitch(uint8_t state) {
    if (state) {
        HAL_GPIO_WritePin(GPIOC, GPIO_PIN_0, GPIO_PIN_SET); // Enable amplifier
    } else {
        HAL_GPIO_WritePin(GPIOC, GPIO_PIN_0, GPIO_PIN_RESET); // Disable amplifier
    }
}

uint16_t ReadVolume(void) {
    // Return volume level 1-65535
    return volume_setting;
}

// In main.c initialization:
PB_StatusTypeDef status = AudioEngine_Init(
    DAC_MasterSwitch,
    ReadVolume,
    MX_I2S2_Init
);

if (status != PB_Idle) {
```

```
printf("Audio engine initialization failed!\n");  
return;  
}  
  
// Now safe to call PlaySample()
```

## `PlaySample()`

Start playback of an audio sample.

```
PB_StatusTypeDef PlaySample(  
    const void *sample_to_play,  
    uint32_t sample_set_sz,  
    uint32_t playback_speed,  
    uint8_t sample_depth,  
    PB_ModeTypeDef mode  
);
```

### Parameters:

- `sample_to_play` : Pointer to audio data in flash/RAM
- `sample_set_sz` : Total samples (all channels combined)
- `playback_speed` : Sample rate in Hz (typically 22000)
- `sample_depth` : 8 or 16 (bits per sample)
- `mode` : `Mode_mono` or `Mode_stereo`

### Returns:

- `PB_Playing` if playback started successfully
- `PB_Error` or `PB_PlayingFailed` on error

### Important Notes:

- Audio data is accessed in real-time during playback (must be in accessible memory)
- DMA directly reads from the provided buffer
- For 16-bit mono audio: `sample_set_sz = num_samples`
- For 16-bit stereo (interleaved): `sample_set_sz = 2 * num_frames`
- For 8-bit mono audio: `sample_set_sz = num_samples`
- For 8-bit stereo (interleaved): `sample_set_sz = 2 * num_frames`
- Blocks briefly while starting DMA
- Configure filters separately with `SetLpf16BitLevel()` or `SetFilterConfig()`

### Example:

```
extern const uint8_t alert_sound_16bit_mono[];  
extern const uint32_t alert_sound_16bit_mono_size;  
  
SetLpf16BitLevel(LPF_Medium); // Medium filtering  
  
PB_StatusTypeDef result = PlaySample(  
    alert_sound_16bit_mono,  
    alert_sound_16bit_mono_size,  
    22000,           // Sample rate  
    16,             // 16-bit  
    Mode_mono  
);
```



```
if (result != PB_Playing) {  
    // Handle error  
    printf("Playback failed: %d\n", result);  
}
```

## `WaitForSampleEnd()`

Block until audio playback completes.

```
PB_StatusTypeDef WaitForSampleEnd(void);
```

### Returns:

- `PB_Idle` when playback finishes
- `PB_Error` if playback was interrupted

### Example:

```
SetLpf16BitLevel(LPF_Soft);  
PlaySample(my_sound, my_sound_size, 22000, 16, Mode_mono);  
WaitForSampleEnd(); // Wait until done  
printf("Playback complete\n");
```

## `PausePlayback()`

Pause ongoing playback (can resume later) with smooth fade-out.

```
PB_StatusTypeDef PausePlayback(void);
```

### Returns:

- `PB_Paused` on success
- `PB_Idle` if no audio was playing

### Notes:

- Pause fadeout duration set by `SetPauseFadeTime()`
- Intelligently handles edge cases to prevent audible artefacts:
- **Pausing during fade-in:** Scales pause fadeout proportionally to start from current volume
- **Pausing during end-of-file fadeout:** Scales pause fadeout to maintain smooth volume continuity
- All transitions use quadratic volume curves for smooth audio

### Example:

```
if (user_pressed_pause_button) {  
    PausePlayback();  
}
```

## `ResumePlayback()`

Resume previously paused audio with smooth fade-in.

```
PB_StatusTypeDef ResumePlayback(void);
```

### Returns:

- `PB_Playing` on success
- `PB_Idle` if no paused audio

### Notes:

- Resume fadein duration set by `SetResumeFadeTime()`
- Audio fades in smoothly from silence using a quadratic volume curve
- Playback resumes from the exact position where it was paused

### Example:

```
if (user_pressed_play_button && prev_state == PB_Paused) {  
    ResumePlayback();  
}
```

---

## Filter Configuration

### `SetFilterConfig()`

Apply a complete filter configuration.

```
void SetFilterConfig(const FilterConfig_TypeDef *cfg);
```

### Parameters:

- `cfg` : Pointer to filter configuration structure

### Example:

```
FilterConfig_TypeDef cfg = {  
    .enable_16bit_biquad_lpf = 1,  
    .enable_soft_dc_filter_16bit = 1,  
    .enable_8bit_lpf = 1,  
    .enable_noise_gate = 0,  
    .enable_soft_clipping = 1,  
    .enable_air_effect = 0,  
    .lpf_makeup_gain_q16 = 70779,  
    .lpf_16bit_level = LPF_Medium,  
    .lpf_16bit_custom_alpha = LPF_16BIT_MEDIUM,  
    .lpf_8bit_level = LPF_Medium,  
    .lpf_8bit_custom_alpha = LPF_MEDIUM  
};  
SetFilterConfig(&cfg);
```

### `GetFilterConfig()`

Read current filter configuration.

```
void GetFilterConfig(FilterConfig_TypeDef *cfg);
```

**Example:**

```
FilterConfig_TypeDef cfg;  
GetFilterConfig(&cfg);  
printf("LPF Level: %d\n", cfg.lpf_16bit_level);
```

**`SetLpf16BitLevel()`**

Change 16-bit LPF aggressiveness (convenience function).

```
void SetLpf16BitLevel(LPF_Level level);
```

**Parameters:**

- **level** : LPF\_VerySoft, LPF\_Soft, LPF\_Medium, LPF\_Firm, or LPF\_Aggressive

**Example:**

```
SetLpf16BitLevel(LPF_Aggressive); // Strong filtering
```

**`SetLpf8BitCustomAlpha()`**

Set custom alpha for the 8-bit one-pole LPF.

```
void SetLpf8BitCustomAlpha(uint16_t alpha);
```

**Example:**

```
SetLpf8BitLevel(LPF_Custom);  
SetLpf8BitCustomAlpha(LPF_MEDIUM);
```

**`GetLpf8BitCustomAlpha()`**

Read the current 8-bit LPF custom alpha value.

```
uint16_t GetLpf8BitCustomAlpha(void);
```

**`CalcLpf8BitAlphaFromCutoff()`**

Compute 8-bit LPF alpha from cutoff and sample rate.

```
uint16_t CalcLpf8BitAlphaFromCutoff(float cutoff_hz, float sample_rate_hz);
```

**Example:**

```
uint16_t alpha = CalcLpf8BitAlphaFromCutoff(4500.0f, 11000.0f);
SetLpf8BitLevel(LPF_Custom);
SetLpf8BitCustomAlpha(alpha);
```

**`SetLpfMakeupGain8Bit()`**

Set post-LPF gain for 8-bit audio.

```
void SetLpfMakeupGain8Bit(float gain);
```

**Parameters:**

- `gain` : Gain multiplier (e.g., 1.0 = no change, 1.08 = +8%)

**Example:**

```
SetLpfMakeupGain8Bit(1.15f); // Boost 8-bit audio by 15%
```

---

**Status Accessors****`GetPlaybackState()`**

Query current playback state (for non-blocking polling).

```
uint8_t GetPlaybackState(void);
```

**Returns:**

- `PB_Idle` , `PB_Playing` , `PB_Paused` , etc.

**Example:**

```
if (GetPlaybackState() == PB_Playing) {
    printf("Audio is playing...\n");
}
```

**`GetPlaybackSpeed()`**

Get current sample rate.

```
uint32_t GetPlaybackSpeed(void);
```

---

**Hardware Integration Functions**

These must be defined by the application to integrate the audio engine with your specific hardware.

## `AudioEngine\_DACSwitch()`

Function pointer to control amplifier GPIO (on/off).

```
extern DAC_SwitchFunc AudioEngine_DACSwitch;

// Application must define:
void MyDACControl(GPIO_PinState setting) {
    if (setting == GPIO_PIN_SET) {
        HAL_GPIO_WritePin(AMP_EN_GPIO_Port, AMP_EN_Pin, GPIO_PIN_SET);    // ON
    } else {
        HAL_GPIO_WritePin(AMP_EN_GPIO_Port, AMP_EN_Pin, GPIO_PIN_RESET);  // OFF
    }
}

// In initialization:
AudioEngine_DACSwitch = MyDACControl;
```

## `AudioEngine\_ReadVolume()`

Function pointer to read volume setting (0–65535). Raw values are scaled as gain during playback. 0 is treated as 1 for minimum volume. Values are subject to the configured volume response curve (linear by default, or non-linear gamma curve if enabled).

```
extern ReadVolumeFunc AudioEngine_ReadVolume;

// Application must define:
uint16_t MyReadVolume(void) {
    // Read GPIO pins or ADC to determine volume level (0-65535)
    uint16_t volume = ReadMyVolumeSource();
    return volume ? volume : 1; // Treat 0 as 1 for minimum volume
}

// In initialization:
AudioEngine_ReadVolume = MyReadVolume;

// Optionally configure volume response curve:
SetVolumeResponseNonlinear(1);    // Enable non-linear response (default)
SetVolumeResponseGamma(2.0f);    // Set gamma exponent (1.0-4.0)
```

## `AudioEngine\_I2SInit()`

Function pointer to re-initialize I2S if needed (called after pause/resume).

```
extern I2S_InitFunc AudioEngine_I2SInit;

// Application must define:
void MyI2SInit(void) {
    MX_I2S2_Init(); // STM32CubeMX-generated initialization
}

// In initialization:
AudioEngine_I2SInit = MyI2SInit;
```

## DMA Callbacks

Connect these to your I2S DMA interrupt handlers:

```
// In your I2S interrupt service routine:
void HAL_I2S_TxHalfCpltCallback(I2S_HandleTypeDef *hi2s) {
    // Called when first half of DMA buffer is transmitted
```

```

    // Audio engine processes next chunk
}

void HAL_I2S_TxCpltCallback(I2S_HandleTypeDef *hi2s) {
    // Called when second half of DMA buffer is transmitted
}

```

The audio engine provides these implementations that will be called automatically.

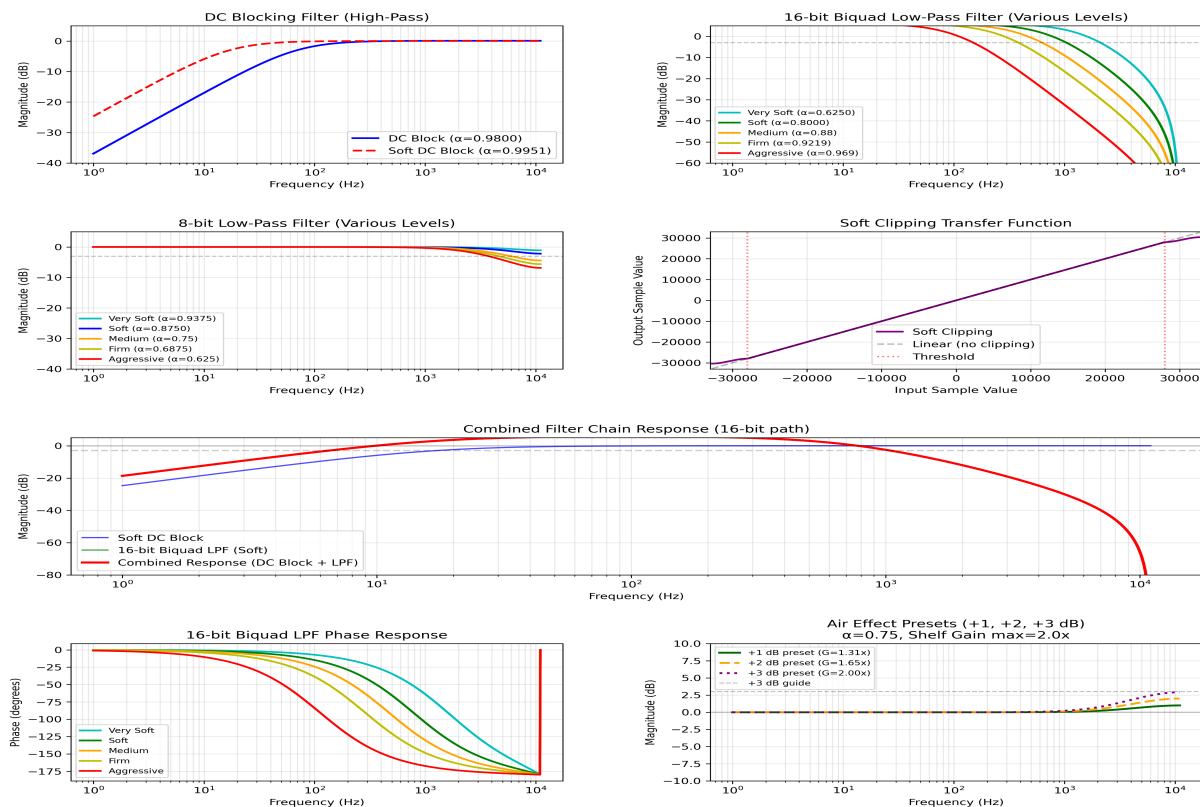
**Interrupt Priority Note:** Ensure `SysTick` has higher priority (numerically lower) than the I2S DMA IRQ. The audio engine may call HAL stop routines from DMA callbacks, and those routines use `HAL_GetTick()` for timeouts. If SysTick cannot preempt the DMA IRQ, timeouts can stall and audio playback can lock up. See `[Core/Inc/stm32g4xx_hal_conf.h](../Core/Inc/stm32g4xx_hal_conf.h)` for `TICK_INT_PRIORITY` and `[Core/Src/main.c](../Core/Src/main.c)` for DMA IRQ priority setup.

---

## Filter Configuration

**Figure 1: Comprehensive Filter Frequency Response Analysis**

**Figure 1: Comprehensive Filter Frequency Response Analysis**  
**Audio Engine DSP Filter Characteristics - Page 1: Filter Responses**  
**STM32 Audio Engine @ 22 kHz Sample Rate**



### 16-bit LPF Aggressiveness Levels

The 16-bit biquad uses **lower  $\alpha$  for heavier filtering** (same direction as the 8-bit one-pole). The coefficient formula `b0 = ((65536 - alpha) * (65536 - alpha)) >> 17` means lower alpha values result in more aggressive low-pass filtering.

Level	Alpha	Notes
Very Soft	0.625	Minimal filtering / brightest tone / highest cutoff
Soft	~0.80	Gentle filtering
Medium	0.875	Balanced filtering
Firm	~0.92	Firm filtering / lower cutoff
Aggressive	~0.97	Strongest filtering / darkest tone / lowest cutoff

- Warm-up (16 passes) still runs to suppress startup artefacts at the most aggressive setting.

#### Recommended Input Range for Best Quality:

The 16-bit biquad has feedback that can cause overshoot and ringing, especially at aggressive filter levels. To avoid clipping while preserving dynamic range:

Level	Recommended Range	Notes
LPF_VerySoft	75–85% of full scale ( $\pm 24,500$ to $\pm 27,750$ )	Minimal overshoot risk
LPF_Soft	70–80% of full scale ( $\pm 22,937$ to $\pm 26,214$ )	Good balance (recommended)
LPF_Medium	70–75% of full scale ( $\pm 22,937$ to $\pm 24,500$ )	Increasing feedback
LPF_Firm	65–75% of full scale ( $\pm 21,300$ to $\pm 24,500$ )	Stronger feedback; moderate headroom
LPF_Aggressive	60–70% of full scale ( $\pm 19,660$ to $\pm 22,937$ )	Strong feedback; conservative headroom essential

#### General Guideline:

Use **70–80% of full scale ( $\pm 23,000$ )** as a safe starting point. If using LPF\_Aggressive, stay closer to 70%; if using LPF\_VerySoft, you can push toward 80–85%.

## 8-bit LPF Aggressiveness Levels

8-bit audio uses a **first-order (one-pole) filter** rather than a biquad. This architecture avoids feedback loop instability on quantized 8-bit data. As a result, the alpha range is narrower than the 16-bit biquad to maintain filter stability.

### Filter Architecture:

- **One-pole formula:**  $\text{output} = (\alpha \times \text{input} + (1 - \alpha) \times \text{prev\_output}) \times \text{makeup\_gain}$
- **Why narrower range:** One-pole filters at low alpha (high filtering) can amplify quantization noise; biquads are more robust to this.

Level	Alpha	Cutoff Freq
Very Soft	0.9375	~3200 Hz
Soft	0.875	~2800 Hz
Medium	0.75	~2300 Hz
Firm	0.6875	~2000 Hz
Aggressive	0.625	~1800 Hz

### Note on Range Differences:

The 16-bit biquad ( $\alpha$ : 0.625  $\rightarrow$  0.97) and 8-bit one-pole ( $\alpha$ : 0.625  $\rightarrow$  0.9375) do *not* span the same range. This is intentional: the biquad's wider range is safe for 16-bit data, while the one-pole's narrower range prevents instability on 8-bit input. Both filters provide LPF\_VerySoft, LPF\_Soft, LPF\_Medium, LPF\_Firm, and LPF\_Aggressive presets for user consistency, but their underlying coefficients differ.

**Important:** The two filter types have the **same relationship** between alpha and filtering: **lower alpha = more filtering** for both architectures. The biquad coefficient formula uses  $(65536 - \alpha)$ , which inverts the typical relationship.

## DC Blocking Filter

Removes DC offset and very low frequencies.

Variant	Alpha	Cutoff Freq	Use
Standard	0.98	~44 Hz	Normal playback
Soft	0.995	~22 Hz	Gentler high-pass (use if ultra-low audio needed)

### When to Enable `enable_soft_dc_filter_16bit` :

- Music with extended bass (< 44 Hz content)
- Subwoofer testing
- Normally leave disabled for typical speech/alerts

## Soft Clipping Threshold

Soft clipping prevents harsh digital distortion when audio peaks exceed a threshold.

### Configuration:



- **Threshold:**  $\pm 28,000$  (85% of  $\pm 32,767$  full scale)
- **Curve:** Cubic smoothstep ( $s(x) = 3x^2 - 2x^3$ )
- **Benefit:** Musical, transparent limiting

**When to Enable:**

- Always recommended (prevents clipping artefacts)
- Disable only if maximum undistorted headroom needed

**Air Effect (High-Shelf Brightening Filter)**

The Air Effect is an optional high-shelf filter that adds presence and brightness to audio by boosting high-frequency content. It uses a simple one-pole shelving architecture for CPU efficiency.

**Configuration (defaults):**

- **Type:** High-shelf one-pole filter
- **Shelf Gain (Q16):** 98304 ( $\sim 1.5\times$ ,  $\approx +1.6$  dB at Nyquist for  $\alpha=0.75$ )
- **Shelf Gain Max (Q16):** 131072 ( $2.0\times$  cap to avoid harshness)
- **Cutoff Alpha:** 0.75 ( $\sim 5\text{--}6$  kHz shelving frequency @ 22 kHz)
- **Default State:** Disabled ( `enable_air_effect = 0` )

**Runtime Control (dB or Q16):**

- `SetAirEffectGainDb(float db)` : set target HF boost in dB (computes Q16 internally, clamped to max)
- `GetAirEffectGainDb(void)` : read current boost in dB
- `SetAirEffectGainQ16(uint32_t gain_q16)` : set raw Q16 shelf gain (clamped)
- `GetAirEffectGainQ16(void)` : read raw Q16 shelf gain
- Presets (built-in): `{+1 dB, +2 dB, +3 dB}` with helpers:
- `SetAirEffectPresetDb(uint8_t preset_index)`
- `CycleAirEffectPresetDb(void)`
- `GetAirEffectPresetIndex/Count/GetAirEffectPresetDb`

**Filter Characteristics:**

The Air Effect works by separating high-frequency content and amplifying it:

1. Extract high-frequency component: `high_freq = input - prev_input`
2. Amplify high frequencies: `boost = high_freq  $\times$  (1 -  $\alpha$ )  $\times$  shelf_gain`
3. Blend with smoothed output: `output = ( $\alpha$   $\times$  input) + ((1 -  $\alpha$ )  $\times$  prev_output) + boost`

**When to Enable:**

- Muffled or dark-sounding samples  $\rightarrow$  adds clarity and presence
- Quiet samples  $\rightarrow$  adds energy and perceived loudness
- Archived audio  $\rightarrow$  brightens aged or compressed recordings
- **Do not enable** if audio already sounds bright or harsh (risk of harshness)

**Typical Use Case:**

```
// Enable Air Effect and choose +2 dB preset
SetAirEffectPresetDb(2); // preset 0=off, 1=+1dB, 2=+2dB, 3=+3dB
// (Auto-disables if preset=0, auto-enables if preset>0)
```

```
SetLpf16BitLevel(LPF_Soft);

PlaySample(
    muffled_doorbell,
    sample_size,
    22000,
    16,
    Mode_mono
);

// Adjust live (e.g., button/UART):
CycleAirEffectPresetDb();
```

### Filter Chain Order:

The Air Effect is positioned after the DC blocking filter but before fade/clipping effects:

```
16/8-bit LPF (optional: enable_16bit_biquad_lpf / enable_8bit_lpf)
↓
DC Blocking Filter (always on: standard or soft mode)
↓
AIR EFFECT (optional: enable_air_effect)
↓
Fade In/Out (always active)
↓
Noise Gate (optional: enable_noise_gate)
↓
Soft Clipping (optional: enable_soft_clipping, recommended)
↓
Volume Scaling (always active)
```

### Tuning the Effect:

- For subtle brightening: use `SetAirEffectGainDb(1.0f)` or `SetAirEffectPresetDb(1)` (+1 dB).
- For more sparkle: use `SetAirEffectGainDb(2.0f)` or `SetAirEffectPresetDb(2)` (+2 dB).
- For stronger presence: `SetAirEffectGainDb(3.0f)` or preset 3 (+3 dB).
- For a subtle lift: `SetAirEffectGainDb(0.0f)` (flat) or reduce gain below 0 dB if adding presence elsewhere.
- For different sample rates (e.g., 48 kHz), raise `AIR_EFFECT_CUTOFF` (higher  $\alpha$ ) to keep the shelf in the upper band.

---

## Playing Audio

### Basic Playback Workflow

```
#include "audio_engine.h"

// 1. Startup (once during initialization, e.g., in main.c)
static void AudioEngine_Init(void) {
    // Wire hardware hooks
    AudioEngine_DACSwitch = DAC_MasterSwitch;
    AudioEngine_ReadVolume = ReadVolume;
    AudioEngine_I2SInit = MX_I2S2_Init;

    // Configure filters
    FilterConfig_TypeDef cfg = filter_cfg; // start from defaults
    cfg.enable_16bit_biquad_lpf = 0;
```

```

    cfg.enable_8bit_lpf          = 1;
    cfg.enable_soft_dc_filter_16bit = 0;
    cfg.enable_soft_clipping     = 1;
    SetFilterConfig(&cfg);

    // Optional tuning
    SetLpf16BitLevel(LPF_Soft);
    SetAirEffectPresetDb(2); // +2 dB preset (auto-enables air effect)
}

// 2. Play an audio sample
static void PlayAlert(void) {
    extern const uint8_t alert_16bit_mono[];
    extern const uint32_t alert_16bit_mono_size;

    PB_StatusTypeDef result = PlaySample(
        alert_16bit_mono,
        alert_16bit_mono_size,
        22000,           // Sample rate
        16,              // 16-bit depth
        Mode_mono        // Mono playback
    );

    if (result == PB_Playing) {
        WaitForSampleEnd();
    }
}

// 3. Non-blocking playback
static void PlayAlertNonBlocking(void) {
    PlaySample(alert_16bit_mono, alert_16bit_mono_size, 22000, 16, Mode_mono);
    // Returns immediately; playback happens in background
}

static void CheckPlaybackStatus(void) {
    if (GetPlaybackState() == PB_Playing) {
        printf("Still playing...\n");
    } else {
        printf("Playback finished\n");
    }
}

```

## Multi-Sample Playback Sequence

```

void PlayDoorbell(void) {
    // First: chime sound (16-bit, gentle filtering)
    SetLpf16BitLevel(LPF_Soft);
    PlaySample(chime_16bit, chime_size, 22000, 16, Mode_mono);
    WaitForSampleEnd();

    // Small delay between sounds
    HAL_Delay(500);

    // Second: bell sound (16-bit, medium filtering)
    SetLpf16BitLevel(LPF_Medium);
    PlaySample(bell_16bit, bell_size, 22000, 16, Mode_mono);
    WaitForSampleEnd();

    printf("Doorbell sequence complete\n");
}

```

## Adjusting Playback on the Fly

```

void InteractivePlayback(void) {
    // Start playback with default settings
    SetLpf16BitLevel(LPF_Medium);
    PlaySample(my_audio, my_audio_size, 22000, 16, Mode_mono);
}

```

```

while (GetPlaybackState() == PB_Playing) {
    // Monitor user input
    if (user_pressed_filter_button) {
        // Change filter level mid-playback
        SetLpf16BitLevel(LPF_Aggressive);
    }

    if (user_pressed_pause_button) {
        PausePlayback();
    }

    if (user_pressed_resume_button) {
        ResumePlayback();
    }

    HAL_Delay(100);
}
}

```

---

## Filter Parameters & Tuning

### Understanding Alpha Coefficients

All filters in the audio engine use first-order or second-order IIR (infinite impulse response) filters with feedback coefficient  $\alpha$  (**alpha**).

#### First-Order Filter:

$$y[n] = \alpha \cdot x[n-1] + (1 - \alpha) \cdot y[n-1]$$

- $\alpha$  close to 1.0: Less filtering (high frequencies pass through)
- $\alpha$  close to 0.0: More filtering (stronger attenuation)

#### Biquad (Second-Order) Filter:

$$y[n] = b_0 \cdot x[n] + b_1 \cdot x[n-1] + b_2 \cdot x[n-2] - a_1 \cdot y[n-1] - a_2 \cdot y[n-2]$$

Where coefficients are derived from  $\alpha$ :

- $b_0 = ((1 - \alpha)^2) / 2$
- $b_1 = 2 \cdot b_0$
- $b_2 = b_0$
- $a_1 = -2 \cdot \alpha$
- $a_2 = \alpha^2$

### Warm-Up Behavior

**Problem:** With aggressive filtering ( $\alpha = 0.625$ ), the first playback sample causes a brief "cracking" sound due to the filter initializing from zero state.

#### Solution: Configurable Warm-Up (Default: 16 passes)

- Automatically invoked when playing 16-bit audio with enabled LPF

- Feeds the first audio sample through the biquad filter `BIQUAD_WARMUP_CYCLES` times on each channel (default: 16)
- Allows filter state to converge smoothly before DMA streaming starts
- Result: Eliminates startup transient artefacts

#### Configuration:

The warm-up behavior can be adjusted by changing the `BIQUAD_WARMUP_CYCLES` define in `audio_engine.h`:

```
#define BIQUAD_WARMUP_CYCLES 16 // Default: 16 passes (was 8)
```

#### Code Example (from `audio_engine.c`):

```
if (sample_depth == 16 && filter_cfg.enable_16bit_biquad_lpf) {
    int16_t first_sample = *((int16_t *)sample_to_play);
    // Run BIQUAD_WARMUP_CYCLES passes to let filter state settle
    for (uint8_t i = 0; i < BIQUAD_WARMUP_CYCLES; i++) {
        ApplyLowPassFilter16Bit(first_sample,
                                &lpf_16bit_x1_left, &lpf_16bit_x2_left,
                                &lpf_16bit_y1_left, &lpf_16bit_y2_left);
        ApplyLowPassFilter16Bit(first_sample,
                                &lpf_16bit_x1_right, &lpf_16bit_x2_right,
                                &lpf_16bit_y1_right, &lpf_16bit_y2_right);
    }
}
```

## Q16 Fixed-Point Arithmetic

All filter coefficients and gains use **Q16 fixed-point representation**:

```
Q16 Value = Integer Value × 65536
Example: 1.0 = 65536 (0x10000)
         0.5 = 32768 (0x8000)
         1.08 ≈ 70779
```

#### Advantages:

- No floating-point hardware required (faster on MCU)
- Deterministic, no rounding surprises
- Easy to implement in assembly if needed

#### Converting Gain to Q16:

```
float gain = 1.08;
uint32_t gain_q16 = (uint32_t)(gain * 65536.0f); // 70779
SetLpfMakeupGain8Bit(gain); // Convenience function
```

## Tuning Guide

### For Speech/Alert Sounds

```
FilterConfig_TypeDef cfg = {
    .enable_16bit_biquad_lpf = 1,
```

```

.enable_soft_dc_filter_16bit = 0, // Not needed for speech
.enable_8bit_lpf = 1,
.enable_noise_gate = 0,          // Or 1 if background noise
.enable_soft_clipping = 1,
.lpf_makeup_gain_q16 = 70779,    // 1.08x
.lpf_16bit_level = LPF_Soft      // Gentle, preserve clarity
};
SetFilterConfig(&cfg);

```

## For Bass-Heavy Music

```

FilterConfig_TypeDef cfg = {
.enable_16bit_biquad_lpf = 1,
.enable_soft_dc_filter_16bit = 1, // Preserve low bass
.enable_8bit_lpf = 1,
.enable_noise_gate = 0,
.enable_soft_clipping = 1,
.lpf_makeup_gain_q16 = 65536,    // 1.0x (no boost)
.lpf_16bit_level = LPF_VerySoft  // Minimal filtering
};
SetFilterConfig(&cfg);

```

## For Noisy Environments

```

FilterConfig_TypeDef cfg = {
.enable_16bit_biquad_lpf = 1,
.enable_soft_dc_filter_16bit = 0,
.enable_8bit_lpf = 1,
.enable_noise_gate = 1,          // Suppress low-level noise
.enable_soft_clipping = 1,
.lpf_makeup_gain_q16 = 70779,
.lpf_16bit_level = LPF_Medium    // Balanced noise reduction
};
SetFilterConfig(&cfg);

```

---

## Volume Control

### Overview

The audio engine applies volume scaling during sample processing. **Your application provides the volume input** by implementing a custom `ReadVolume()` function that returns a `uint16_t` value in the range 1–65535.

You can implement this function using various input methods:

- **Digital GPIO inputs** (OPT1–OPT3 pins for 3-bit binary encoding) — example pattern provided below
- **Analog ADC input** (12-bit potentiometer or variable resistance) — example pattern provided below
- **Other methods** (CAN bus, UART commands, network, etc.) — follow the same scaling approach

All implementations support **non-linear (logarithmic) volume response** to match human hearing perception.

## Non-Linear Volume Response

Human hearing perceives loudness logarithmically, not linearly. The audio engine provides a configurable gamma-curve response.

Enable in main.h:

```
#define VOLUME_RESPONSE_NONLINEAR // Enable non-linear curve
#define VOLUME_RESPONSE_GAMMA 2.0f // Gamma exponent (typical value)
```

How it works:

- Linear input (1–65535) is normalized to 0.0–1.0
- Gamma curve is applied: `output = input^(1/gamma)`
- Result scales back to 1–65535 for audio attenuation

Gamma Values:

Gamma	Perception	Use Case
1.0	Linear	Reference (no curve)
2.0	Quadratic (recommended)	Most intuitive for human control
2.5	Stronger curve	Aggressive low-volume response

With **gamma = 2.0** (quadratic):

- **Low volumes** (0–50%): Small slider movement → big loudness change
- **High volumes** (50–100%): Big slider movement → small loudness change
- Result: More intuitive volume "feel" matching human perception

To disable and use linear scaling:

```
//#define VOLUME_RESPONSE_NONLINEAR // Comment this out
```

## Volume Control Implementation

> **Note:** Volume input implementation is application-specific. The audio engine simply invokes the `ReadVolume()` callback and expects a `uint16_t` value in the range 1–65535. The application is responsible for:

- > - Choosing the volume input method (digital GPIO, ADC potentiometer, CAN bus, UART command, etc.)
- > - Scaling that input to the 1–65535 range
- > - Handling debouncing, hysteresis, or other input conditioning
- > - Being aware that if `VOLUME_RESPONSE_NONLINEAR` is enabled, the engine applies a gamma curve to the returned value

### Application Example: Digital GPIO Volume Control

3-bit binary selector example (shows the pattern for application developers):

Selection	Binary	Scaled to 1–65535
Maximum	0b000	65535
75%	0b001	49151
50%	0b010	32767
...	...	...
Minimum	0b111	1

#### Example implementation in application (main.c):

```
uint16_t ReadVolume(void) {
    // Application reads three GPIO pins for volume
    uint8_t v =
        ( ( (OPT3_GPIO_Port->IDR & OPT3_Pin) != 0 ) << 2 ) |
        ( ( (OPT2_GPIO_Port->IDR & OPT2_Pin) != 0 ) << 1 ) |
        ( ( (OPT1_GPIO_Port->IDR & OPT1_Pin) != 0 ) << 0 );

    v = 7 - v; // Invert so 0b000 = max volume
    uint32_t scaled = ( (uint32_t)v * 65535U ) / 7U; // Map 0-7 to 1-65535
    return (uint16_t)scaled;
}
```

## Application Example: Analog ADC Volume Control

### Potentiometer example (shows the pattern for application developers):

#### Example implementation in application (main.c):

```
uint16_t ReadVolume(void) {
    // Application reads 12-bit ADC (0-4095)
    // Scale to 1-65535 range
    uint32_t lin = ((uint32_t)adc_raw * 65535U) / 4095U;
    return (uint16_t)lin;
}

void HAL_ADC_ConvCpltCallback( ADC_HandleTypeDef *hadc )
{
    if( hadc == &hadc1 ) {
        adc_raw = HAL_ADC_GetValue( &hadc1 );
    }
}
```

---

## Hardware Integration

### STM32CubeMX Configuration

#### 1. I2S Setup (e.g., I2S2 or other available I2S peripheral):

- Mode: Master Transmit Only
- Sample Rate: 22000 Hz
- Data Format: 16-bit, Mono or Stereo



- DMA: Enable DMA for I2Sxext\_TX (or similar based on board)
- ## 2. DMA Configuration:
- Mode: Circular
  - Word Width: Word (32-bit)
  - Enable both **Half-Transfer Complete** and **Transfer Complete** interrupts
- ## 3. GPIO:
- Amplifier enable pin (e.g., PE7 on STM32G474, or any available GPIO)
  - Volume select pins (2–3 GPIO inputs for 3-level selector)
  - LED indicators (optional)

## Application Integration Template

```
#include "audio_engine.h"
#include "stm32g4xx_hal.h"

/* Hardware control functions (application-specific) */
void DAC_MasterSwitch(GPIO_PinState setting) {
    HAL_GPIO_WritePin(AMP_EN_GPIO_Port, AMP_EN_Pin, setting);
}

uint16_t ReadVolume(void) {
    // Application-specific: read volume from GPIO, ADC, UART, etc.
    // Must return 1-65535 (0 is treated as 1)
    // The audio engine applies non-linear response if enabled

    // Example: 3-bit GPIO selector
    uint8_t v =
        ( ( (OPT3_GPIO_Port->IDR & OPT3_Pin) != 0 ) << 2 ) |
        ( ( (OPT2_GPIO_Port->IDR & OPT2_Pin) != 0 ) << 1 ) |
        ( ( (OPT1_GPIO_Port->IDR & OPT1_Pin) != 0 ) << 0 );
    v = 7 - v; // Invert: 0b000 = max volume
    uint32_t scaled = ( (uint32_t)v * 65535U ) / 7U;
    return (uint16_t)scaled;
}

/* Main initialization (in main.c HAL_Init sequence) */
void SystemInit_Audio(void) {
    // Set hardware callbacks before playing audio
    AudioEngine_DACSwitch = DAC_MasterSwitch;
    AudioEngine_ReadVolume = ReadVolume;
    AudioEngine_I2SInit = MX_I2S2_Init;

    // Configure filter settings (optional, defaults work for most cases)
    FilterConfig_TypeDef cfg;
    GetFilterConfig(&cfg);
    cfg.enable_soft_clipping = 1;
    SetFilterConfig(&cfg);

    printf("Audio engine ready\n");
}

/* DMA interrupt handlers (in stm32g4xx_it.c or similar) */
void I2S2_IRQHandler(void) {
    HAL_I2S_IRQHandler(&hi2s2);
}

/* HAL weak function overrides */
void HAL_I2S_TxHalfCpltCallback(I2S_HandleTypeDef *hi2s) {
    if (hi2s->Instance == I2S2) {
        // Audio engine handles this internally
    }
}

void HAL_I2S_TxCpltCallback(I2S_HandleTypeDef *hi2s) {
    if (hi2s->Instance == I2S2) {
        // Audio engine handles this internally
    }
}
```

```
}  
}
```

---

## Examples

### Example 1: Simple Doorbell

```
#include "audio_engine.h"  
  
// Audio data (define in flash)  
extern const uint8_t doorbell_mono_16bit[];  
extern const uint32_t doorbell_mono_16bit_size;  
  
void PlayDoorbell(void) {  
    SetLpf16BitLevel(LPF_Soft);  
    PB_StatusTypeDef result = PlaySample(  
        doorbell_mono_16bit,  
        doorbell_mono_16bit_size,  
        22000,           // 22 kHz  
        16,             // 16-bit  
        Mode_mono       // Mono  
    );  
  
    if (result == PB_Playing) {  
        WaitForSampleEnd();  
        printf("Doorbell complete\n");  
    } else {  
        printf("Failed to play doorbell\n");  
    }  
}
```

### Example 2: Multi-Tone Alert with Different Filters

```
void PlayAlert(void) {  
    extern const uint8_t tone1[], tone2[], tone3[];  
    extern const uint32_t tone1_size, tone2_size, tone3_size;  
  
    // First tone: gentle  
    SetLpf16BitLevel(LPF_VerySoft);  
    PlaySample(tone1, tone1_size, 22000, 16, Mode_mono);  
    WaitForSampleEnd();  
    HAL_Delay(200);  
  
    // Second tone: medium  
    SetLpf16BitLevel(LPF_Medium);  
    PlaySample(tone2, tone2_size, 22000, 16, Mode_mono);  
    WaitForSampleEnd();  
    HAL_Delay(200);  
  
    // Third tone: aggressive (emphasis)  
    SetLpf16BitLevel(LPF_Aggressive);  
    PlaySample(tone3, tone3_size, 22000, 16, Mode_mono);  
    WaitForSampleEnd();  
}
```

### Example 3: Voice Message with Configurable Filtering

```
void PlayVoiceMessage(LPF_Level filter_level) {  
    extern const uint8_t message_16bit[];
```

```

extern const uint32_t message_16bit_size;

// Set filter before playback
SetLpf16BitLevel(filter_level);

PB_StatusTypeDef result = PlaySample(
    message_16bit,
    message_16bit_size,
    22000,
    16,
    Mode_mono
);

if (result == PB_Playing) {
    printf("Message playing with filter level %d\n", filter_level);
}

// Usage:
// PlayVoiceMessage(LPF_Soft);          // Clear
// PlayVoiceMessage(LPF_Aggressive);    // Compressed

```

## Example 4: Pause/Resume Functionality

```

volatile uint8_t pause_requested = 0;

void PlaybackTask(void) {
    SetLpf16BitLevel(LPF_Medium);
    PlaySample(my_audio, my_audio_size, 22000, 16, Mode_mono);

    while (GetPlaybackState() == PB_Playing) {
        if (pause_requested) {
            PausePlayback();
            printf("Paused\n");

            while (!resume_requested && GetPlaybackState() == PB_Paused) {
                HAL_Delay(50);
            }

            ResumePlayback();
            printf("Resumed\n");
            pause_requested = 0;
            resume_requested = 0;
        }

        HAL_Delay(50);
    }
}

// Button handler:
void EXTI_PauseButton_Handler(void) {
    pause_requested = 1;
}

void EXTI_ResumeButton_Handler(void) {
    resume_requested = 1;
}

```

## Example 5: Non-Blocking Playback with Status Checking

```

void NonBlockingPlayback(void) {
    // Start playing
    SetLpf16BitLevel(LPF_VerySoft);
    PlaySample(background_music, bg_music_size, 22000, 16, Mode_stereo);

    // Do other work while audio plays
    for (int i = 0; i < 100; i++) {
        if (GetPlaybackState() == PB_Playing) {

```

```

    printf("Playing: %d%% complete\n", (i+1));
} else {
    printf("Playback finished\n");
    break;
}

HAL_Delay(100); // 10 second total wait
}
}

```

## Example 6: Accessibility — Filter Settings for Hard of Hearing

This example demonstrates filter configuration optimized for users with hearing loss, emphasizing speech clarity and presence without over-filtering.

```

void SetAccessibleAudio(void) {
    // Configuration optimized for hearing-impaired listeners
    // Focus: speech clarity and presence in 2-6 kHz band
    FilterConfig_TypeDef cfg = {
        .enable_16bit_biquad_lpf = 1,
        .lpf_16bit_level = LPF_Soft,           // Gentle filtering preserves clarity
        .enable_soft_dc_filter_16bit = 1,      // Softer DC removal (22 Hz cutoff)
        .enable_8bit_lpf = 1,
        .enable_noise_gate = 0,                // Keep quiet consonants (s, th, sh)
        .enable_soft_clipping = 1,            // Reduce harsh peaks
        .enable_air_effect = 1,                // Boost presence in 2-6 kHz
        .lpf_makeup_gain_q16 = 82000           // ~1.25x gain (Q16 fixed-point)
    };

    SetFilterConfig(&cfg);
    SetAirEffectPresetDb(2);                  // +2 dB presence boost (mid-range)
}

// Usage in doorbell application
void play_accessible_alert(void) {
    SetAccessibleAudio();

    PlaySample(alert_tone, alert_size, 22000, 16, Mode_mono);
    WaitForSampleEnd();
}

```

### Design Rationale:

- **LPF\_Soft** ( $\alpha \approx 0.80$ ) — Gentler than Medium/Aggressive; prevents over-smoothing that muddies speech
- **No Noise Gate** — Preserves subtle consonants and quiet speech details critical for comprehension
- **Air Effect +2 dB** — Compensates for typical age-related high-frequency loss; boosts presence band (2–6 kHz where speech consonants live)
- **Makeup Gain ~1.25x** — Offsets the LPF attenuation, maintaining perceived loudness
- **Soft DC Filter** — Gentler transition than standard DC blocking; avoids unnatural clicks

### Alternative for Severe Loss:

```

// For users with more pronounced loss, use Very Soft + stronger presence
cfg.lpf_16bit_level = LPF_VerySoft;        // Lightest filtering
SetFilterConfig(&cfg);
SetAirEffectPresetDb(3);                   // +3 dB (strongest preset)

```

---

## Troubleshooting

### Issue: Audio Not Playing

**Symptoms:** `PlaySample()` returns `PB_Error` or `PB_PlayingFailed`

**Solutions:**

1. Verify I2S2 is initialized via `MX_I2S2_Init()`
2. Check that `AudioEngine_I2SInit` callback is set
3. Confirm DMA is enabled for I2S2 TX
4. Check amplifier GPIO is working: `HAL_GPIO_WritePin(AMP_EN_GPIO_Port, AMP_EN_Pin, GPIO_PIN_SET)`
5. Verify audio data pointer is valid (in flash or accessible RAM)

**Debug:**

```
uint8_t state = GetPlaybackState();
printf("Playback state: %d\n", state); // 0=Idle, 2=Playing, etc.

SetLpf16BitLevel(LPF_Soft);
PB_StatusTypeDef result = PlaySample(my_audio, my_size, 22000, 16, Mode_mono);
printf("PlaySample result: %d\n", result);
```

### Issue: Audio Playing but Distorted or Crackling

**Symptoms:** Output contains harsh noise or crackles, especially at start

**Solutions:**

**1. Enable soft clipping:**

```
FilterConfig_TypeDef cfg;
GetFilterConfig(&cfg);
cfg.enable_soft_clipping = 1;
SetFilterConfig(&cfg);
```

**2. Adjust filter level to reduce aggressive processing:**

```
SetLpf16BitLevel(LPF_Medium); // Instead of LPF_Aggressive
```

**3. Reduce volume:**

- Check that hardware volume setting (GPIO or analog input) is at reasonable level
- Audio data may have peaks at or near  $\pm 32,767$

**4. Verify audio data:**

- Check that 16-bit samples are properly formatted (little-endian on ARM)
- Ensure sample rate matches playback speed

### Issue: Filter Sounds Too Thin/Bright

**Symptoms:** High frequencies dominate, lacks bass/warmth

**Solutions:**

```
// Increase filtering aggressiveness
SetLpf16BitLevel(LPF_Aggressive);

// Or enable soft DC blocking for extended low end
FilterConfig_TypeDef cfg;
GetFilterConfig(&cfg);
cfg.enable_soft_dc_filter_16bit = 1; // 22 Hz instead of 44 Hz
SetFilterConfig(&cfg);
```

### Issue: Filter Sounds Too Muffled/Dull

**Symptoms:** High frequencies are suppressed, loss of clarity

**Solutions:**

```
// Reduce filtering aggressiveness
SetLpf16BitLevel(LPF_VerySoft);

// Or disable some filters entirely
FilterConfig_TypeDef cfg;
GetFilterConfig(&cfg);
cfg.enable_8bit_lpf = 0; // If playing 16-bit audio
SetFilterConfig(&cfg);
```

### Issue: 8-bit Audio Sounds Noisy

**Symptoms:** Audible quantization noise from 8-bit samples

**Solutions:**

1. **Enable TPDF dithering (automatic)** - should already be on by default
2. **Increase makeup gain:**

```
SetLpfMakeupGain8Bit(1.15f); // Boost by 15%
```

3. **Use 16-bit audio if available** - much better quality

### Issue: Memory Corruption / Hard Fault

**Symptoms:** Microcontroller resets or freezes during audio playback

**Common Causes:**

1. **Audio buffer pointer is invalid or not in accessible memory**

- Audio data must be in flash (constant) or main RAM
- Not in stack-only RAM

2. **DMA configuration issue**

- Verify DMA word width is 32-bit (not 8 or 16)
- Check DMA direction is I2S TX (transmit)

3. **I2S interrupt interfering with audio processing**

- Ensure `HAL_I2S_IRQHandler()` is called in ISR
- Verify DMA interrupt priorities don't conflict

- **SysTick must have higher priority (numerically lower) than the I2S DMA IRQ.** HAL stop routines invoked from DMA callbacks rely on `HAL_GetTick()` timeouts, which can stall if SysTick cannot preempt the DMA IRQ.

Debug:

```
// Add validation before playing
if ((uint32_t)audio_ptr < 0x08000000 && (uint32_t)audio_ptr >= 0x0A000000) {
    printf("Invalid audio pointer: 0x%08X\n", (uint32_t)audio_ptr);
}
```

---

Performance Notes

CPU Load

- **Processing overhead:** < 5% @ 22 kHz with full filter chain enabled
- **Per-sample time:** ~50 CPU cycles for all filters
- **DMA-driven:** Most processing offloaded from main CPU loop

Memory Usage

```
Flash (.text/.rodata, Release): ~12.9 KB (audio engine + filters + Air Effect prese
ts)
RAM: ~2.5 KB (state variables + playback buffer)
```

Latency

- **Playback latency:** ~93 ms (2048 samples @ 22 kHz)
- 50 ms for DMA buffer
- 43 ms for warm-up and initial processing
- **Pause/Resume:** Immediate (within one DMA block, ~45 ms)

Quality Metrics

Metric	Value	Notes
Sample Rate	22 kHz	Nyquist: 11 kHz
Bit Depth	16-bit (native)	8-bit with TPDF dithering
Dynamic Range	96 dB (16-bit)	48 dB (8-bit effective)
SNR (w/ dithering)	102 dB (8-bit)	TPDF reduces quantization noise
THD (soft clipping)	< 0.1%	Cubic smoothstep minimizes distortion

Power Consumption

- **STM32G474 (typical):** ≤40 mA (core + peripherals)

- **I2S + DMA Active:** ~10 mA additional
- **Amplifier (MAX98357A):** ~100 mA @ 0.5W output, >500 mW capable
- **Total System @ 0.5W audio:** ~150 mA @ 5V
- **Total System @ 1W audio:** ~200 mA @ 5V

**Note:** The MAX98357A amplifier can deliver over 500 mW to an 8Ω speaker, significantly exceeding the STM32's typical current draw.

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## Summary

The Audio Engine provides a complete, production-ready solution for embedded audio playback with professional DSP filtering. Key design principles:

1. **Fixed-Point Integer Math** - No FPU required, deterministic on MCU
2. **Modular Filter Chain** - Enable/disable each stage independently
3. **Runtime Configuration** - Adjust filter parameters without recompilation
4. **DMA-Driven Streaming** - Efficient background audio playback
5. **Warm-Up Initialization** - Eliminates startup artefacts with aggressive filtering

For questions or issues, refer to the troubleshooting section or review the provided code examples.

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**Document Version:** 1.0

**Last Updated:** 2026-01-24

**Audio Engine Version:** Modularized with Widened 16-bit LPF & Warm-Up Support