

# **Audio Engine**

## **User Manual**

STM32 DSP Audio Playback System  
for microcontrollers with I2S support  
Version 2.0

8-bit | 16-bit | Mono | Stereo | Runtime DSP | No FPU

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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 bytes = ~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, // Size in bytes
    22000, // Sample rate (Hz)
    16, // Bit depth (16 = 16-bit)
    Mode_stereo, // Stereo playback
    LPF_Soft // Low-pass filter level
);

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

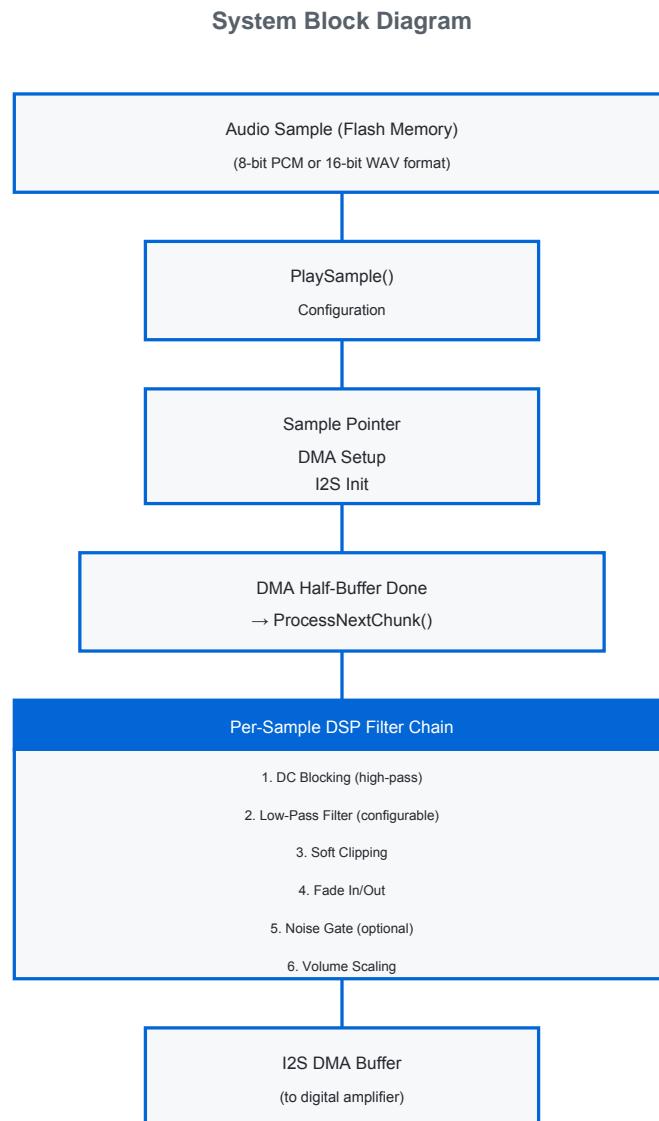
### 3. Configure Filters

```
// 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.

### 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. Biquad Low-Pass Filter (*Optional - enable\_8bit\_lpf*)

- Same architecture as 16-bit path
- Separate aggressiveness levels for 8-bit audio
- Cutoff range (22 kHz fs, approx): ~2.6 kHz (Aggressive), ~1.7 kHz (Medium), ~0.9 kHz (Soft), ~0.4 kHz (Very Soft)

## 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 biquad filters.

```
typedef enum {
    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_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 biquad LPF
    uint8_t enable_noise_gate; // Enable/disable noise gate
    uint8_t enable_soft_clipping; // Enable/disable soft clipping
    uint32_t lpf_makeup_gain_q16; // Post-LPF gain in Q16 fixed-point
    LPF_Level lpf_16bit_level; // 16-bit LPF aggressiveness
} 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

### `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

### 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  
    }  
}
```

```

    }

}

uint8_t ReadVolume(void) {
    // Return volume level 0-2
    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,
    LPF_Level lpf_level
);

```

### Parameters:

- `sample_to_play` : Pointer to audio data in flash/RAM
- `sample_set_sz` : Total size in bytes (not samples)
- `playback_speed` : Sample rate in Hz (typically 22000)
- `sample_depth` : 8 or 16 (bits per sample)
- `mode` : `Mode_mono` or `Mode_stereo`
- `lpf_level` : LPF aggressiveness (LPF\_VerySoft to LPF\_Aggressive)

### 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 = 2 * num_samples`
- For 16-bit stereo audio: `sample_set_sz = 4 * num_samples` (if interleaved L/R)
- Blocks briefly while starting DMA

**Example:**

```
extern const uint8_t alert_sound_16bit_mono[];

extern const uint32_t alert_sound_16bit_mono_size;

PB_StatusTypeDef result = PlaySample(
    alert_sound_16bit_mono,
    alert_sound_16bit_mono_size,
    22000, // Sample rate
    16, // 16-bit
    Mode_mono,
    LPF_Medium // Medium filtering
);

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:**

```
PlaySample(my_sound, my_sound_size, 22000, 16, Mode_mono, LPF_Soft);
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,  
    .lpf_makeup_gain_q16 = 70779,  
    .lpf_16bit_level = 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
```

### `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 (1–255). Values are scaled as a linear gain; 0 is treated as 1.

```

extern ReadVolumeFunc AudioEngine_ReadVolume;

// Application must define:

uint8_t MyReadVolume(void) {
    // Read GPIO pins or ADC to determine volume level (1-255)
    uint8_t volume = ReadMyVolumeSource();

    return volume ? volume : 1; // Ensure minimum volume of 1
}

// In initialization:
AudioEngine_ReadVolume = MyReadVolume;

```

## `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.

---

## Filter Configuration

### 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
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_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:** `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 → 0.97) and 8-bit one-pole ( $\alpha$ : 0.625 → 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:** ±28,000 (85% of ±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 (~1.5x,  $\approx +1.6$  dB at Nyquist for  $\alpha=0.75$ )
- **Shelf Gain Max (Q16):** 131072 (2.0x cap to avoid harshness)
- **Cutoff Alpha:** 0.75 (~5–6 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 × (1 - α) × shelf_gain`
3. Blend with smoothed output: `output = (α × input) + ((1 - α) × prev_output) + boost`

### When to Enable:

- Muffled or dark-sounding samples → adds clarity and presence
- Quiet samples → adds energy and perceived loudness
- Archived audio → 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)

PlaySample(
    muffled_doorbell,
    sample_size,
```

```

22000,
16,
Mode_mono,
LPF_Soft
);

// 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)

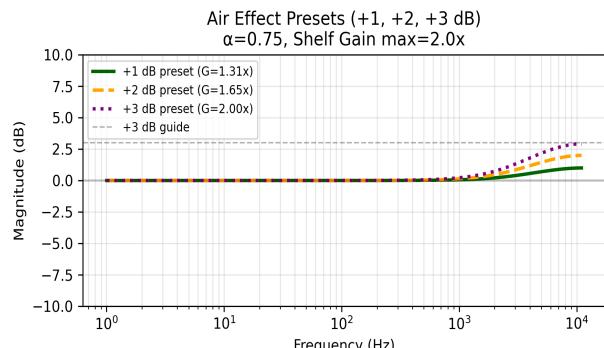
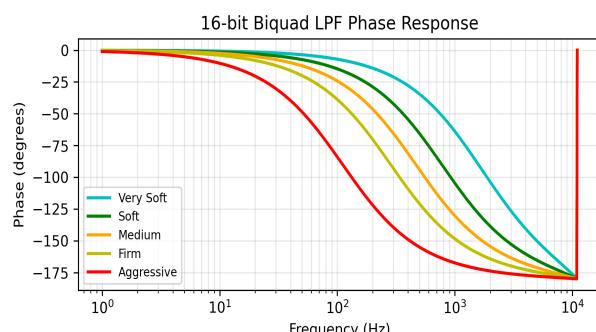
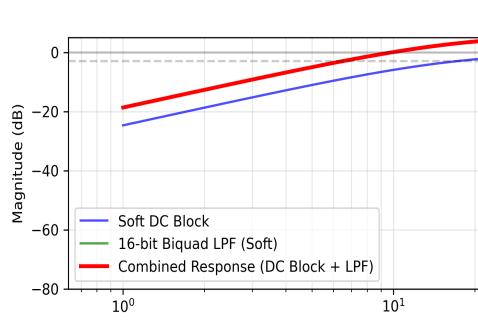
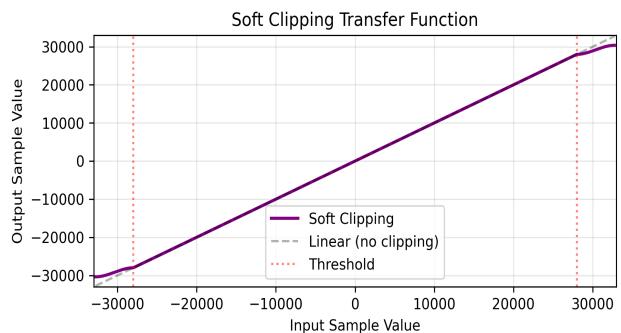
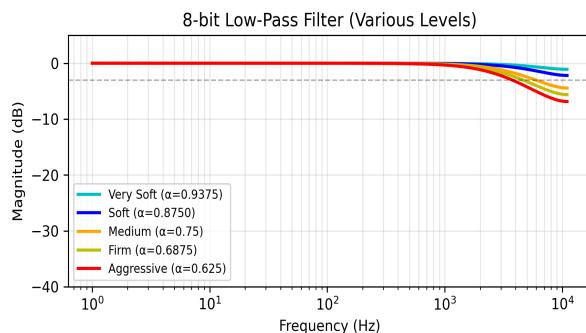
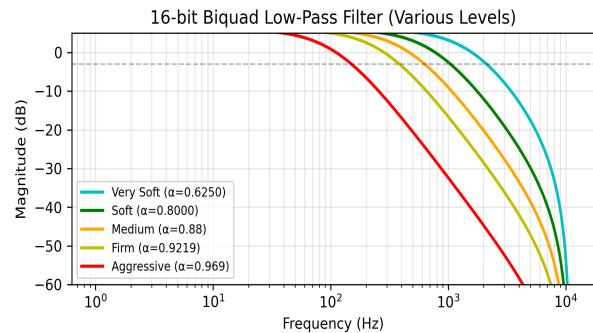
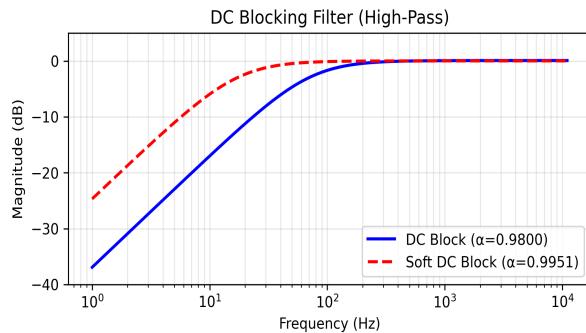
```

**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.

---

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



## 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
        LPF_Soft // Gentle filtering
    );

    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, LPF_Soft);
}

// Returns immediately; playback happens in background
}

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

## Multi-Sample Playback Sequence

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

    // Small delay between sounds
    HAL_Delay(500);

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

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

## Adjusting Playback on the Fly

```

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

    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 * x[n-1] + (1 - alpha) * 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] = b0*x[n] + b1*x[n-1] + b2*x[n-2] - a1*y[n-1] - a2*y[n-2]
```

Where coefficients are derived from  $\alpha$ :

- $b0 = ((1 - \alpha)^2) / 2$
- $b1 = 2 \cdot b0$

- $b_2 = b_0$
- $a_1 = -2 \cdot \alpha$
- $a_2 = \alpha^2$

## 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 x 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. Volume can be controlled via:

1. **Digital GPIO inputs** (OPT1–OPT3 pins for 3-bit binary encoding)
2. **Analog ADC input** (12-bit potentiometer or variable resistance)

Both pathways 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 (0–255) is normalized to 0.0–1.0
- Gamma curve is applied: `output = input^(1/gamma)`
- Result scales back to 0–255 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
```

## Digital Volume Control (GPIO-Based)

### Pin Configuration:

- **OPT1** (LSB): GPIO input pin
- **OPT2**: GPIO input pin
- **OPT3** (MSB): GPIO input pin

### Encoding (3-bit binary, inverted so 0b000 = max volume):

OPT3	OPT2	OPT1	Level	Effective Volume
0	0	0	Maximum	100% (255/255)
0	0	1	75%	191/255
0	1	0	50%	127/255
0	1	1	37%	95/255
1	0	0	25%	63/255
1	0	1	19%	47/255
1	1	0	12%	31/255
1	1	1	Minimum	1/255

### Implementation:

```
#define VOLUME_INPUT_DIGITAL // Enable in audio_engine.h

// In ReadVolume() (main.c):
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
uint16_t scaled = ( (uint16_t)v * 255U ) / 7U; // Map 0-7 to 0-255
return scaled ? (uint8_t)scaled : 1U; // Return 1-255
```

## Analog Volume Control (ADC-Based)

### Pin Configuration:

- Connect potentiometer to **ADC1 Channel X** (e.g., PA0)
- 12-bit ADC reads 0–4095 from wiper voltage

#### Implementation:

```
// Disable VOLUME_INPUT_DIGITAL in audio_engine.h

// In ReadVolume() (main.c):

uint16_t lin = adc_raw / 16; // Scale 0-4095 down to ~0-255

if( lin > 220 ) lin = 220; // Cap to 220 to avoid clipping
return lin ? lin : 1; // Return 1-255
```

#### ADC Interrupt Handler:

```
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 */

void DAC_MasterSwitch(GPIO_PinState setting) {
    HAL_GPIO_WritePin(AMP_EN_GPIO_Port, AMP_EN_Pin, setting);
}

uint8_t ReadVolume(void) {
    // Read three GPIO pins for 3-bit digital volume (1-8 scaled to 1-255)
    // Or read an analog input via ADC
    // The audio engine will apply non-linear (logarithmic) response
    // if VOLUME_RESPONSE_NONLINEAR is enabled in main.h

#ifdef VOLUME_INPUT_DIGITAL
    // Digital: Use OPT1-OPT3 GPIO pins
    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
    uint16_t scaled = ( (uint16_t)v * 255U ) / 7U; // Map 0-7 to 0-255
    return scaled ? (uint8_t)scaled : 1U;
#else
    // Analog: Use 12-bit ADC
    uint16_t lin = adc_raw / 16; // Scale to 0-255 range
    if( lin > 220 ) lin = 220;
    return lin ? lin : 1;
#endif
}

/* 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)
    FilterConfigTypeDef 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);
    ...
    ...
}

```

## 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) {
    PB_StatusTypeDef result = PlaySample(
        doorbell_mono_16bit,
        doorbell_mono_16bit_size,
        22000, // 22 kHz
        16, // 16-bit
        Mode_mono, // Mono
        LPF_Soft // Gentle filtering
    );

    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
    PlaySample(tone1, tone1_size, 22000, 16, Mode_mono, LPF_VerySoft);
    WaitForSampleEnd();
    HAL_Delay(200);

    // Second tone: medium
    PlaySample(tone2, tone2_size, 22000, 16, Mode_mono, LPF_Medium);
    WaitForSampleEnd();
    HAL_Delay(200);

    // Third tone: aggressive (emphasis)
    PlaySample(tone3, tone3_size, 22000, 16, Mode_mono, LPF_Aggressive);
    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,
        filter_level // Use same level
    );

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

    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
    PlaySample(background_music, bg_music_size, 22000, 16, Mode_stereo, LPF_VerySoft)
    ;

    // 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, LPF_Soft);

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.

PB_StatusTypeDef result = PlaySample(my_audio, my_size, 22000, 16, Mode_mono, LPF_Sof
t);

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 ±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

**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 presets
)
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