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`

Generated: February 03, 2026

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

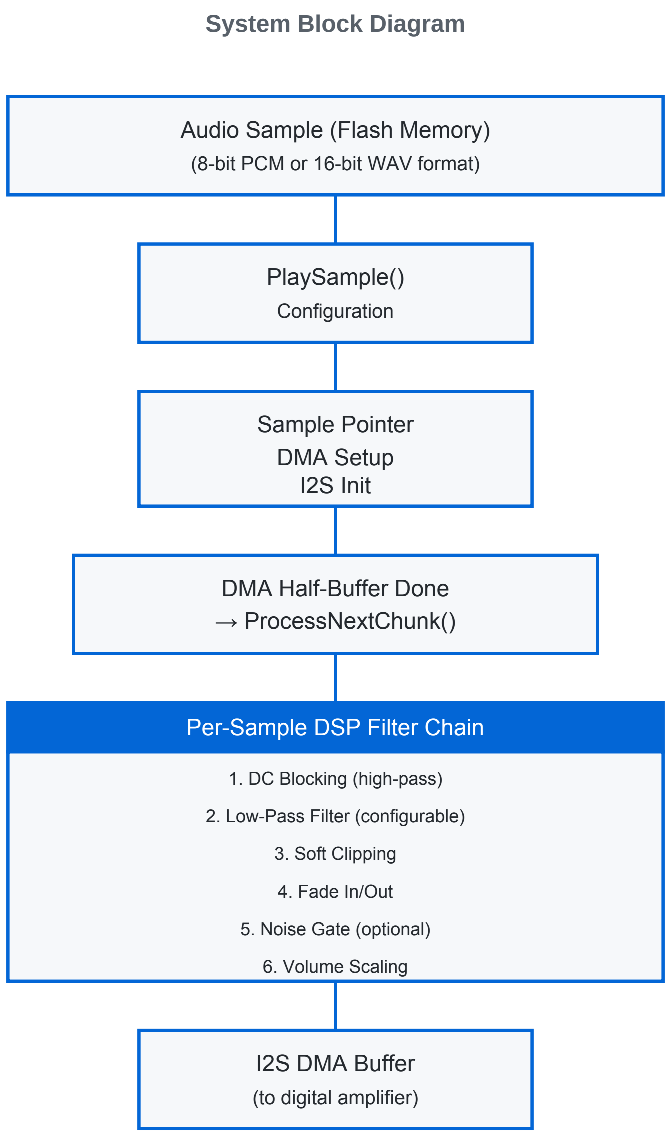
```
// Assuming 'doorbell_sound' is a 16-bit mono WAV sample in flash memory
// 44,100 bytes = ~2 seconds @ 22 kHz, 16-bit mono

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 ±512 amplitude
   - Suppresses quantization noise during silence
   - Disabled by default

6. **Soft Clipping** *(Optional - enable_soft_clipping)*
   - Smooth cubic curve limiting above ±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 (α = 0.625)
```

```
 LPF_Soft, // Gentle filtering (α ≈ 0.80)
 LPF_Medium, // Balanced filtering (α = 0.875)
 LPF_Firm, // Firm filtering (α ≈ 0.92)
 LPF_Aggressive // Strong filtering (α ≈ 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 α 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 (±24,500 to ±27,750)	Minimal overshoot risk
LPF_Soft	70–80% of full scale (±22,937 to ±26,214)	Good balance (recommended)
LPF_Medium	70–75% of full scale (±22,937 to ±24,500)	Increasing feedback
LPF_Aggressive	60–70% of full scale (±19,660 to ±22,937)	Strong feedback; conservative headroom essential

General Guideline:

Use **70–80% of full scale (±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 = (α × input + (1 – α) × prev_output) × 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 (α: 0.625 → 0.97) and 8-bit one-pole (α: 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 (~1.5×, ≈ +1.6 dB at Nyquist for α=0.75)
- **Shelf Gain Max (Q16)**: 131072 (2.0× 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:

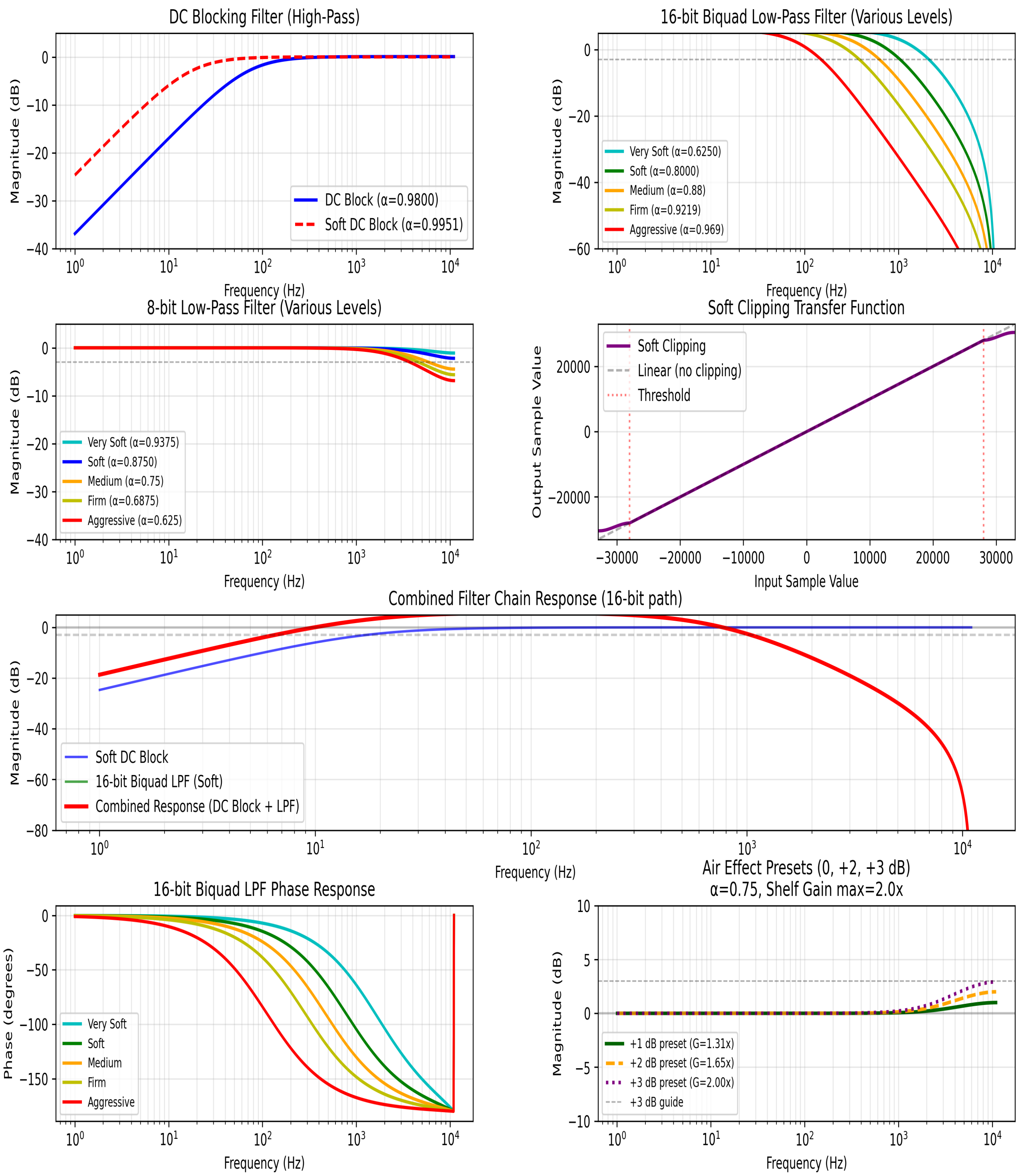
The Air Effect is positioned after the DC blocking filter but before fade/clipping effects:

```
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 more sparkle: use `SetAirEffectGainDb(2.0f)` or `SetAirEffectPresetDb(1)` (+2 dB).
- For stronger presence: `SetAirEffectGainDb(3.0f)` or preset 2 (+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 α) 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_Sof
 t);
 // Returns immediately; playback happens in background
}

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

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 α **(alpha)**.

First-Order Filter:

```
y[n] = α · x[n-1] + (1 - α) · y[n-1]
```

- α close to 1.0: Less filtering (high frequencies pass through)
- α 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 α:

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

- $b2 = b0$
- $a1 = -2 \cdot \alpha$
- $a2 = \alpha^2$

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. 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)
 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);
... (truncated)
```

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** (α ≈ 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.

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.

Document Version: 1.0

Last Updated: 2026-01-24

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