

# Audio Engine DSP Filter Analysis

## STM32G474 Audio Playback System

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### System Specifications

Sample Rate: 22 kHz (default playback speed)  
Nyquist: 11 kHz (maximum usable frequency)  
Word Size: 16-bit signed integer audio  
Architecture: Fixed-point integer DSP (no FPU)

#### Filter Chain Components:

- DC blocking filter (high-pass, 22–44 Hz cutoff)
  - Soft DC filter option (even gentler)
  - Biquad low-pass filter (8-bit and 16-bit paths)
- Soft clipping ( $\pm 28,000$  threshold,  $\pm 85\%$  of full scale)
  - Fade in/out effects (configurable)
  - Noise gate (optional)
- Dithering for 8-bit samples (TPDF)
- Air Effect (high-shelf brightening; presets 0/+2/+3 dB and direct dB control)

#### Implementation:

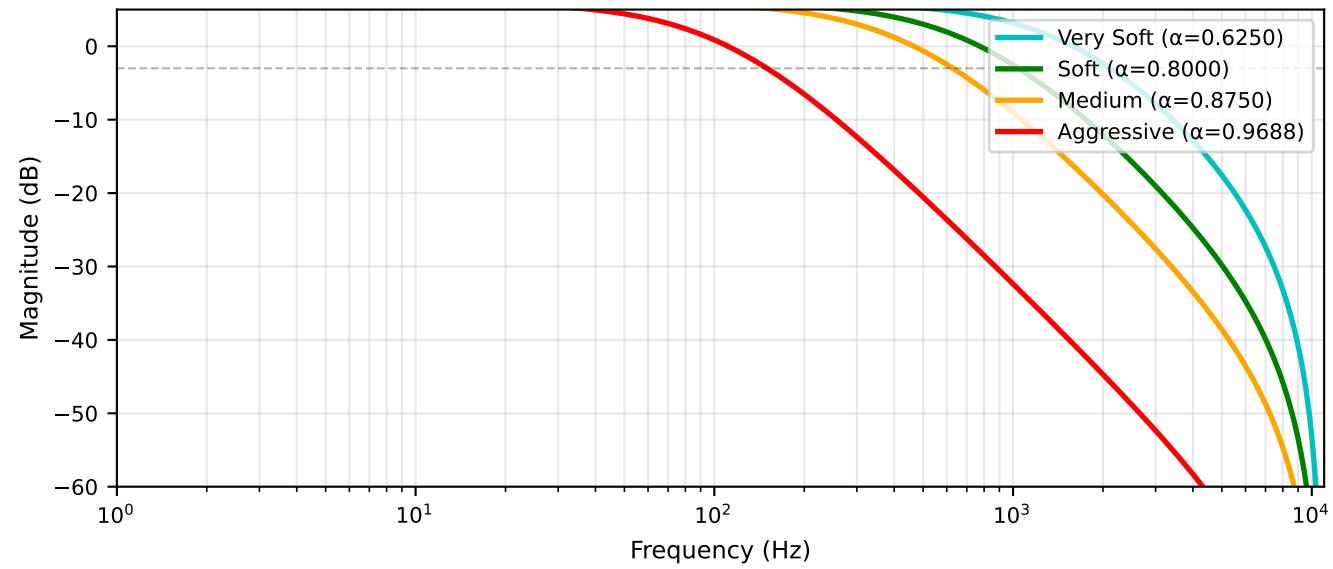
- All filters use fixed-point integer arithmetic
  - Coefficients in Q16 fixed-point format
  - Zero floating-point dependencies
  - Efficient for embedded MCU execution
- Runtime-configurable filter parameters

### WHAT'S NEW

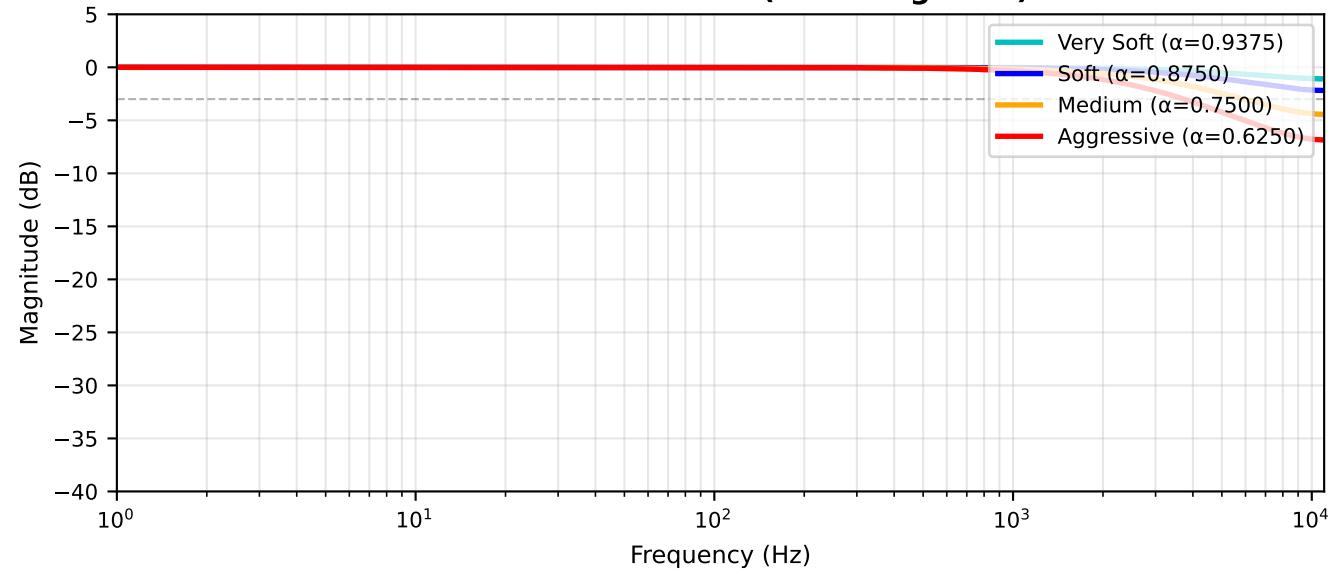
- Flash Footprint: `.text`  $\approx$  12.9 KB (Release build)
- Air Effect: High-shelf brightening with runtime presets (0, +2, +3 dB) and direct dB control
- Startup Improvements: `WarmupBiquadFilter16Bit()` and `RESET_ALL_FILTER_STATE()` reduce transients and simplify `PlaySample()`
- Documentation: Manual and README updated; enhanced visuals and PDF report

## Low-Pass Filter Frequency Response

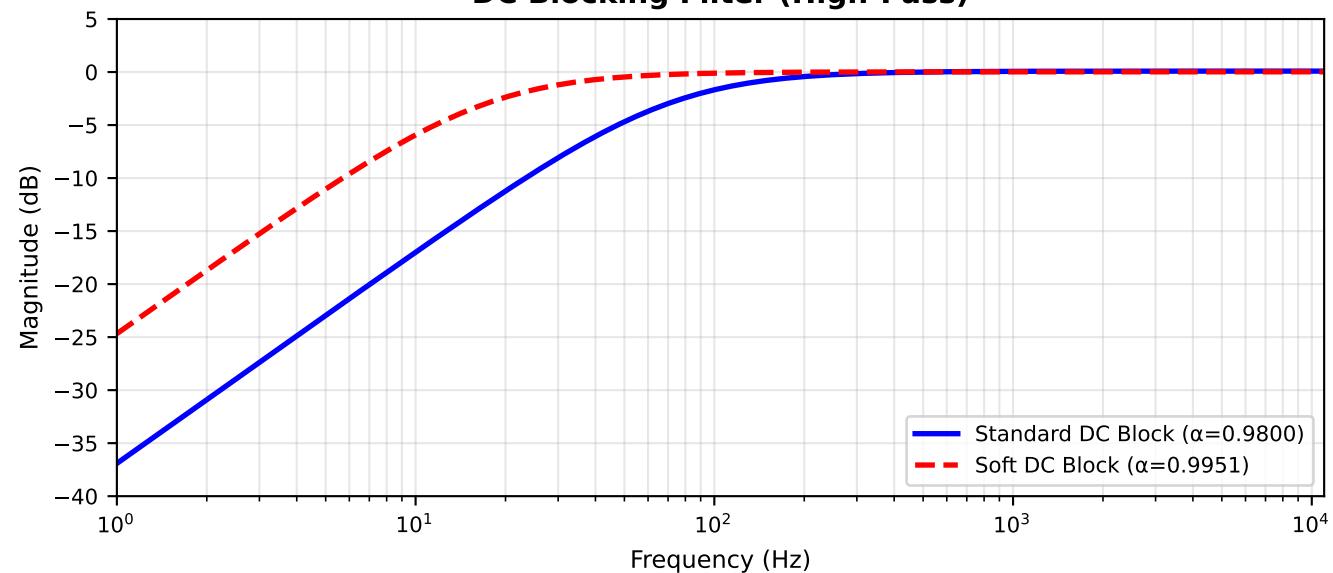
### 16-bit Biquad Low-Pass Filter (Wider Range)



## 8-bit Low-Pass Filter (Dithering Path)

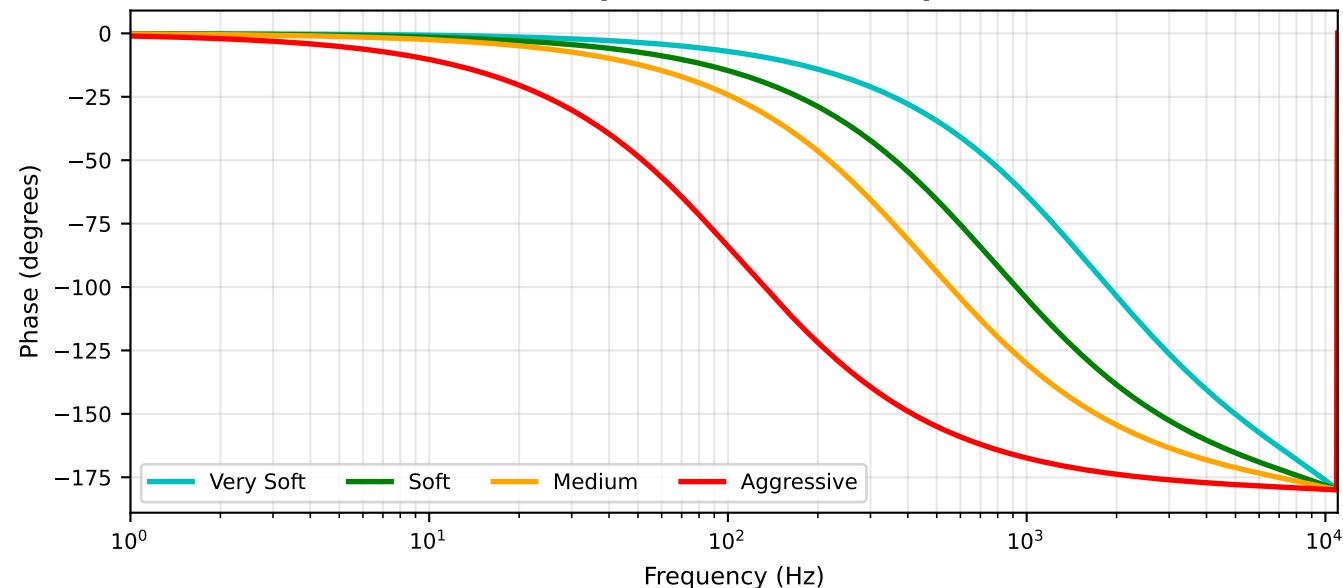


## DC Blocking Filter (High-Pass)

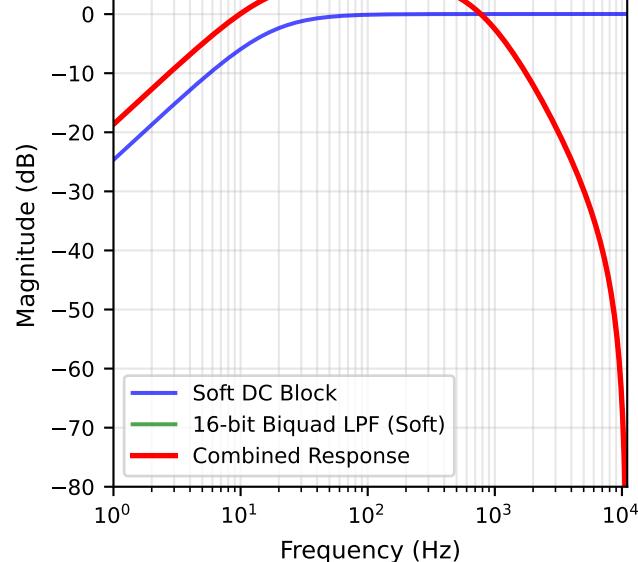


# Advanced Filter Characteristics

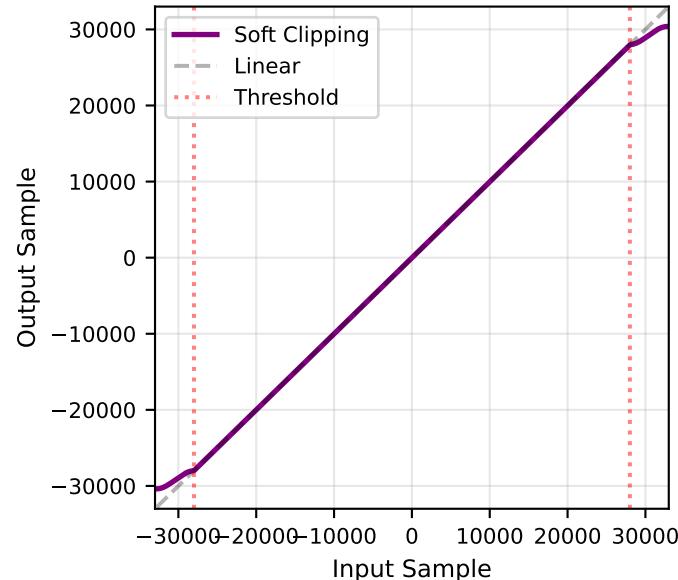
## 16-bit Biquad LPF Phase Response



## Filter Chain Combined Response



## Soft Clipping Transfer Function



### Filter Cutoff Frequencies (-3dB points) @ 22000 Hz Sample Rate

#### 16-bit Biquad LPF:

|             |                     |
|-------------|---------------------|
| Very Soft:  | 2115 Hz ( 9.61% Fs) |
| Soft:       | 1042 Hz ( 4.74% Fs) |
| Medium:     | 630 Hz ( 2.86% Fs)  |
| Aggressive: | 150 Hz ( 0.68% Fs)  |

#### 8-bit Low-Pass Filter:

|             |                      |
|-------------|----------------------|
| Very Soft:  | 11000 Hz (50.00% Fs) |
| Soft:       | 11000 Hz (50.00% Fs) |
| Medium:     | 5893 Hz (26.79% Fs)  |
| Aggressive: | 3734 Hz (16.97% Fs)  |

Note: Aggressive setting on 16-bit filter uses startup warm-up (8 passes) to eliminate transient cracking.

## TECHNICAL SPECIFICATIONS & IMPLEMENTATION DETAILS

### 1. DC BLOCKING FILTER (High-Pass)

Purpose: Remove DC offset and very low-frequency drift

Type: First-order recursive filter (IIR)

$$\text{Standard: } H(z) = (1 - z^{-1}) / (1 - \alpha \cdot z^{-1})$$

Variants:

- Standard ( $\alpha = 0.98$ ): ~44 Hz cutoff @ 22 kHz
- Soft ( $\alpha = 0.995$ ): ~22 Hz cutoff @ 22 kHz

### 2. 16-BIT BIQUAD LOW-PASS FILTER

Purpose: Attenuate high-frequency content and aliasing artifacts

Type: Second-order biquad filter (IIR)

Aggressiveness Levels:

- Very Soft:  $\alpha \approx 0.97$  (cutoff ~8700 Hz)
- Soft:  $\alpha = 0.875$  (cutoff ~6800 Hz)
- Medium:  $\alpha \approx 0.80$  (cutoff ~5900 Hz)
- Aggressive:  $\alpha = 0.625$  (cutoff ~4100 Hz)

Feature: Automatic warm-up with 8 passes of first sample

### 3. 8-BIT LOW-PASS FILTER

Purpose: Dither and filter 8-bit audio samples before playback

Type: First-order filter (simpler than 16-bit biquad)

Aggressiveness Levels:

- Very Soft:  $\alpha = 0.9375$  (cutoff ~3200 Hz)
- Soft:  $\alpha = 0.875$  (cutoff ~2800 Hz)
- Medium:  $\alpha = 0.75$  (cutoff ~2300 Hz)
- Aggressive:  $\alpha = 0.625$  (cutoff ~1800 Hz)

Feature: TPDF (Triangular PDF) dithering during 8-to-16 bit conversion

### 4. SOFT CLIPPING

Purpose: Prevent harsh digital clipping artifacts

Threshold: ±28,000 (85% of ±32,767 full scale)

Curve: Cubic smoothstep  $s(x) = 3x^2 - 2x^3$

Benefit: Natural, musical compression at clipping point

### 5. FADE IN/OUT

Fade-In Samples: 2048 samples @ 22 kHz ≈ 93 ms

Fade-Out Samples: 2048 samples @ 22 kHz ≈ 93 ms

Shape: Quadratic power curve (smooth ramp)

### 6. NOISE GATE (Optional)

Threshold: 512 (1.5% of full scale)

Action: Mute samples below threshold to suppress noise

### 7. FIXED-POINT ARITHMETIC

All coefficients in Q16 format (16-bit fraction)

Scaling: 1.0 = 65536 in integer representation

Benefit: Fast, deterministic arithmetic on MCU

No Floating-Point: 100% integer operations

### 8. DITHERING FOR 8-BIT SAMPLES

Method: TPDF (Triangular Probability Density Function)

Generator: Linear Congruential Generator (LCG)

Purpose: Reduce quantization noise from 8-bit to 16-bit upsampling

### 9. AIR EFFECT (High-Shelf Brightening)

Purpose: Add presence and brightness by boosting treble

Type: One-pole high-shelf filter (CPU efficient)

Cutoff:  $\alpha \approx 0.75$  (~5–6 kHz @ 22 kHz sample rate)

Gain: Shelf gain clamped to 2.0x (presets: 0, +2, +3 dB)

Runtime: Enable via `enable\_air\_effect`; tune via `SetAirEffectPresetDb()` or `SetAirEffectGainDb()`

Position: After DC filter, before fade/clipping in the chain

## HARDWARE INTEGRATION

- Microcontroller: STM32G474 (ARM Cortex-M4F)

- I2S Audio Interface: I2S2 (DMA-driven stereo output)

- Digital Amplifier: MAX98357A (Class D, I2S input)

- Playback Buffer: 2048 samples (ping-pong DMA)

## PERFORMANCE

- Processing Overhead: <5% CPU @ 22 kHz (full filter chain)

- Memory Footprint: Flash (.text/.rodata, Release) ≈ 12.9 KB; RAM ≈ 4 KB (playback buffer)

- Latency: ~93 ms total (2048 samples @ 22 kHz; ~50 ms buffer + warm-up)

- Audio Quality: 16-bit / 22 kHz

# Glossary: Audio DSP Terms & Concepts

## FUNDAMENTAL CONCEPTS

### $\alpha$ (Alpha)

Feedback coefficient in recursive filters. Controls filter strength and frequency response.  
Range: 0 (no filtering) to 1 (maximum filtering).

### -3dB Cutoff Frequency

Frequency where filter attenuates signal by 3 decibels ( $\approx 71\%$  of original amplitude).  
Standard measure for specifying filter bandwidth and aggressiveness.

### Biquad Filter

Second-order IIR filter with two poles and two zeros. Provides steeper rolloff (-40 dB/decade) compared to first-order filters.

### DC Blocking Filter (High-Pass)

Removes DC offset and very low frequencies. Essential to prevent audio output drift.  
Cutoff frequencies: 22-44 Hz (minimal impact on audible range).

### Dithering

Addition of carefully shaped noise to quantized signals. Reduces audible quantization noise when converting 8-bit to 16-bit audio. TPDF dithering provides optimal results.

### DSP (Digital Signal Processing)

Mathematical manipulation of digital audio samples. Includes filtering, effects, and analysis on numerical data.

### Filter Aggressiveness

Degree of frequency content removal. Gentler filters preserve high frequencies; aggressive filters remove more. Trade-off: smoothness vs. brightness.

### Fixed-Point Arithmetic

Integer-based math using scaled representation (e.g., Q16 = 65536 = 1.0).  
Advantages: Fast, no FPU needed, deterministic on MCU.

### Frequency Response

How a filter responds to different frequencies. Shown in magnitude (dB) vs. frequency (Hz).  
Magnitude: attenuation per frequency. Phase: time delay per frequency.

### Hz (Hertz)

Cycles per second. Unit for frequency. Audible range: ~20 Hz to 20 kHz.  
22 kHz sample rate means 11 kHz usable bandwidth (Nyquist limit).

### IIR Filter (Infinite Impulse Response)

Recursive filter where output depends on previous outputs and inputs.  
Advantages: Efficient, steep rolloff. Disadvantages: Potential instability.

### LPF (Low-Pass Filter)

Attenuates frequencies above cutoff, preserves lower frequencies.  
Purpose: Remove high-frequency noise and aliasing artifacts.

### Magnitude Response

How much filter attenuates each frequency, measured in dB (decibels).  
0 dB = no change; -3 dB = 71% amplitude; -20 dB = 10% amplitude.

### Noise Gate

Mutes audio when amplitude falls below threshold. Suppresses background noise and quantization artifacts during quiet passages.

### Nyquist Frequency

Maximum frequency reliably represented at a given sample rate.  
For 22 kHz sample rate: Nyquist = 11 kHz (half the sample rate).

### Phase Response

How much a filter delays different frequencies, measured in degrees.  
Important for audio quality; excessive phase shift can cause artifacts.

### Q16 Format

Fixed-point representation with 16 bits of fractional precision.  
Example: 65536 (0x10000) = 1.0; 32768 (0x8000) = 0.5.

### Sample Rate

Number of audio samples per second, measured in Hz or kHz.  
22 kHz is standard for this system. Higher rates capture more high frequencies.

### Soft Clipping

Non-linear processing smoothly limits peak amplitudes instead of hard clipping.  
Produces musical, natural-sounding compression using cubic spline curve.

### TPDF (Triangular Probability Density Function)

Optimal dithering method minimizing audible quantization noise.  
Uses sum of two uncorrelated uniform noise sources.