

Audio Engine DSP Filter Analysis

STM32G474 Audio Playback System

Report Generated: 2026-01-25 22:03:31

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:

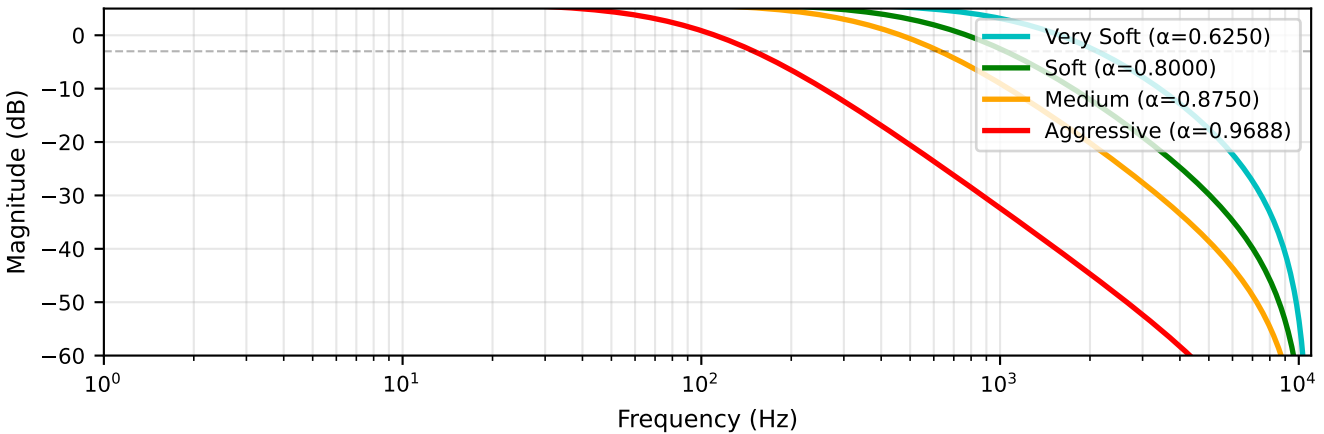
- All filters use fixed-point integer arithmetic
 - Coefficients in Q16 fixed-point format
 - Zero floating-point dependencies
 - Efficient for embedded MCU execution
 - Runtime-configurable filter parameters

WHAT'S NEW

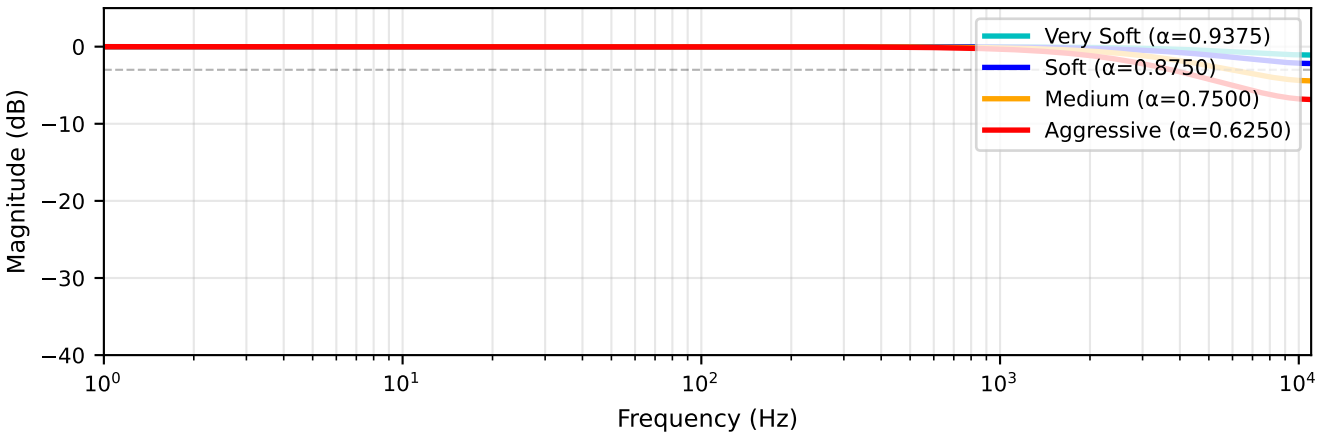
- Flash Footprint: .text \approx 12.9 KB (Release build)
- Air Effect: High-shelf brightening with runtime presets (0, +2, +3 dB) and direct dB control
- Startup Improvements: WarmupBiquadFilter16Bit() and RESET_ALL_FILTER_STATE() reduce transients and simplify PlaySample()
 - Documentation: Manual and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

Filter Frequency Response

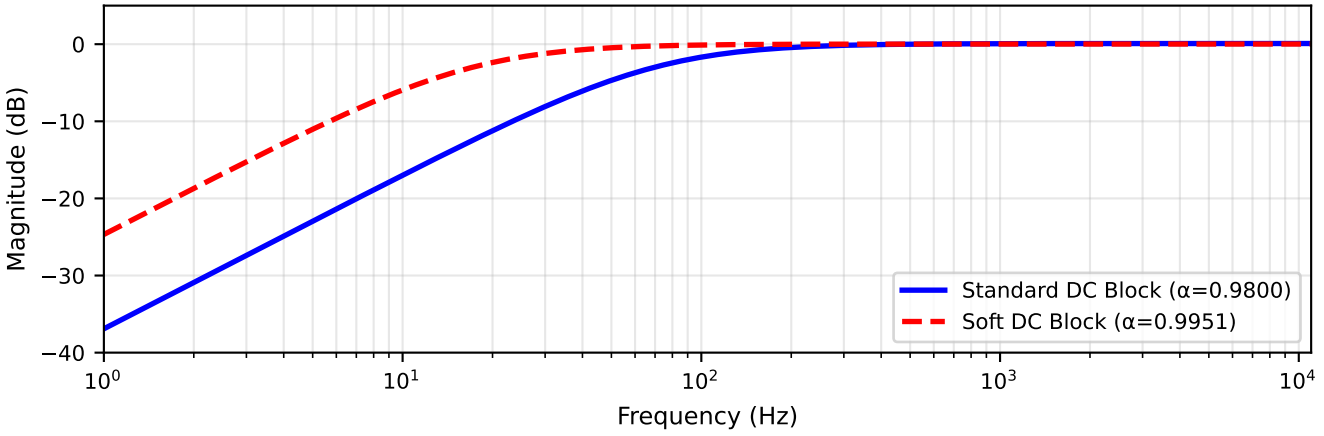
16-bit Biquad Low-Pass Filter (Wider Range)



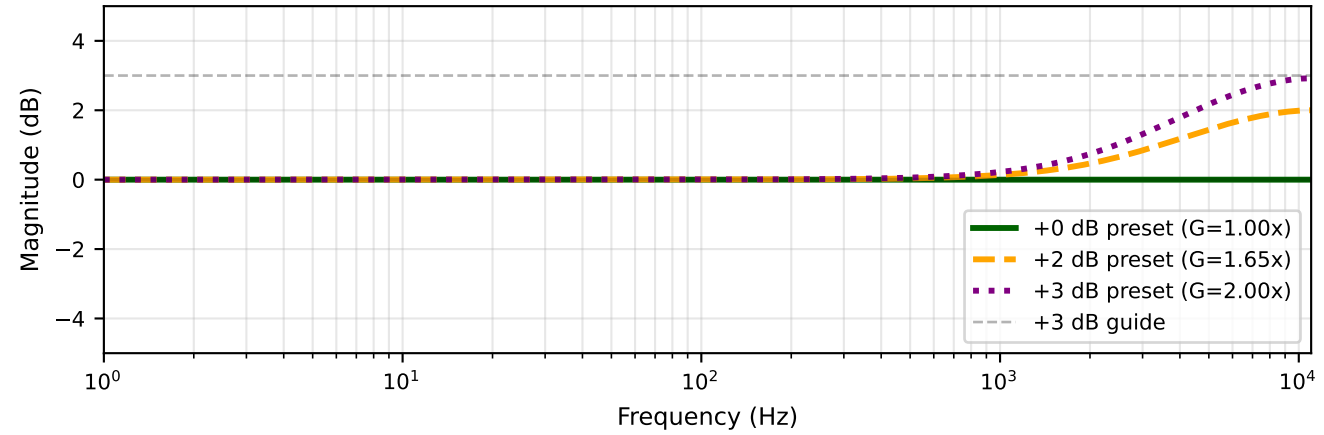
8-bit Low-Pass Filter (Dithering Path)



DC Blocking Filter (High-Pass)

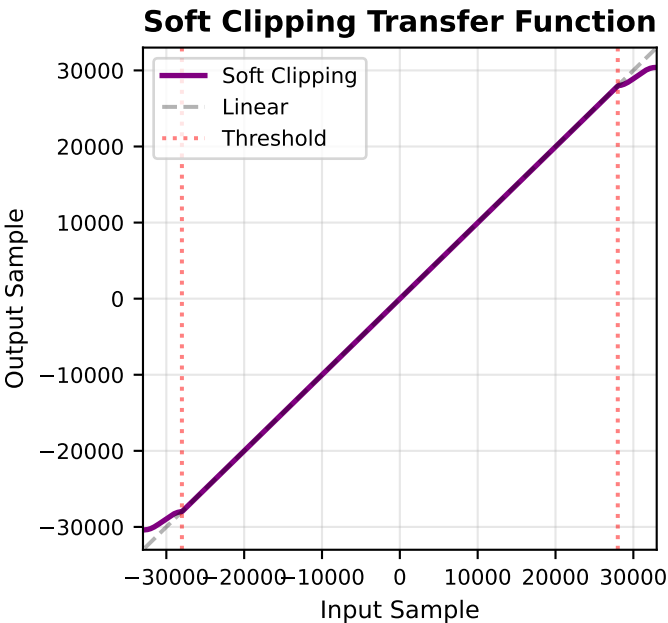
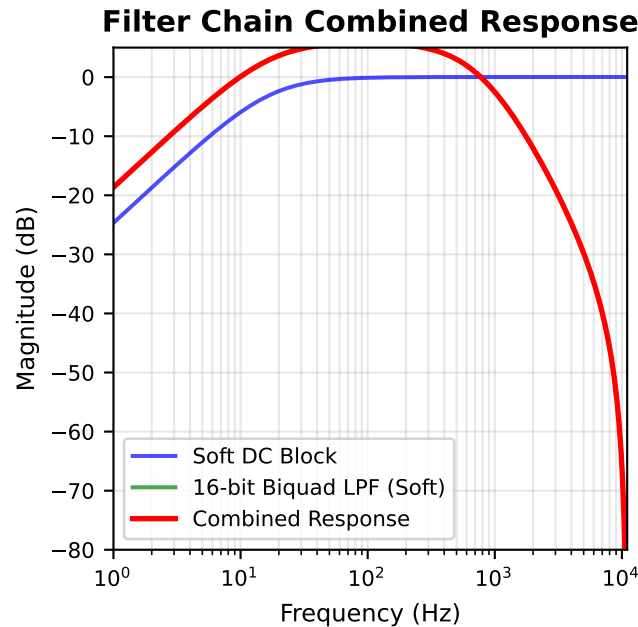
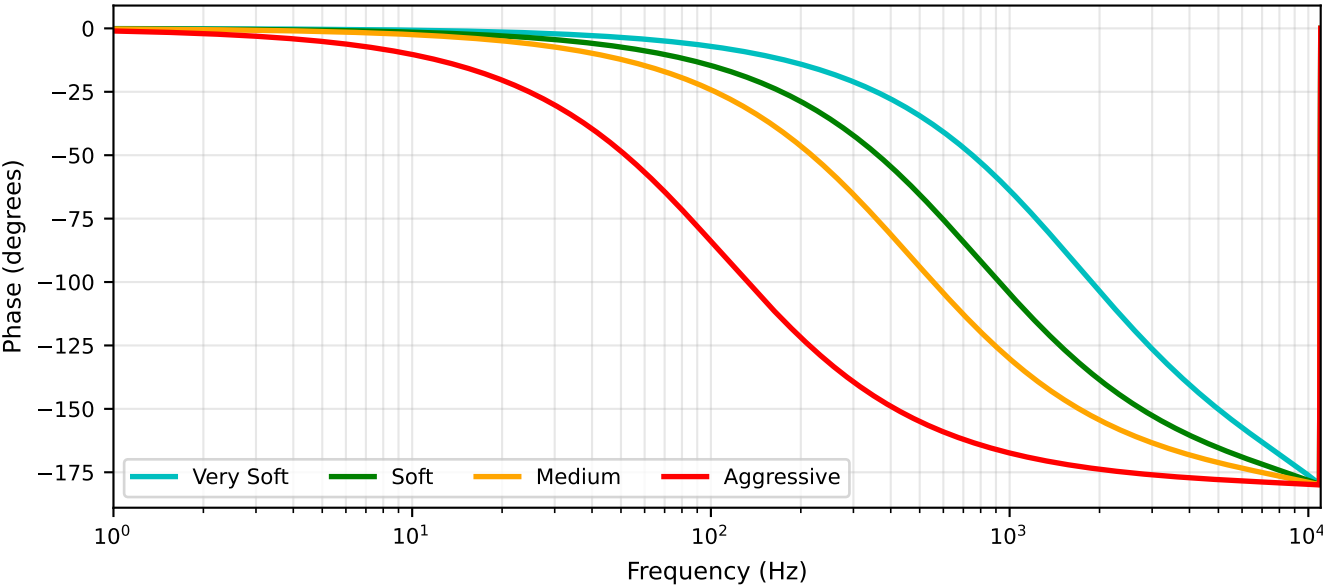


Air Effect Presets (0, +2, +3 dB) - High-Shelf Brightening



Advanced Filter Characteristics

16-bit Biquad LPF Phase Response



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 (16 passes, configurable) 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)

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: $\alpha \approx 0.875$ (cutoff ~6800 Hz)
- Medium: $\alpha \approx 0.80$ (cutoff ~5900 Hz)
- Aggressive: $\alpha \approx 0.625$ (cutoff ~4100 Hz)

Feature: Automatic warm-up with 16 passes (configurable) 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: $\pm 28,000$ (85% of $\pm 32,767$ 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 \approx 93 ms

Fade-Out Samples: 2048 samples @ 22 kHz \approx 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: $\alpha \approx 0.75$ ($\approx 5-6$ kHz @ 22 kHz sample rate)

Gain: Shelf gain clamped to $2.0\times$ (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) \approx 12.9 KB; RAM \approx 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

α (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 random noise sources.