Audio Engine DSP Filter Analysis

STM32G474 Audio Playback System

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System Specifications

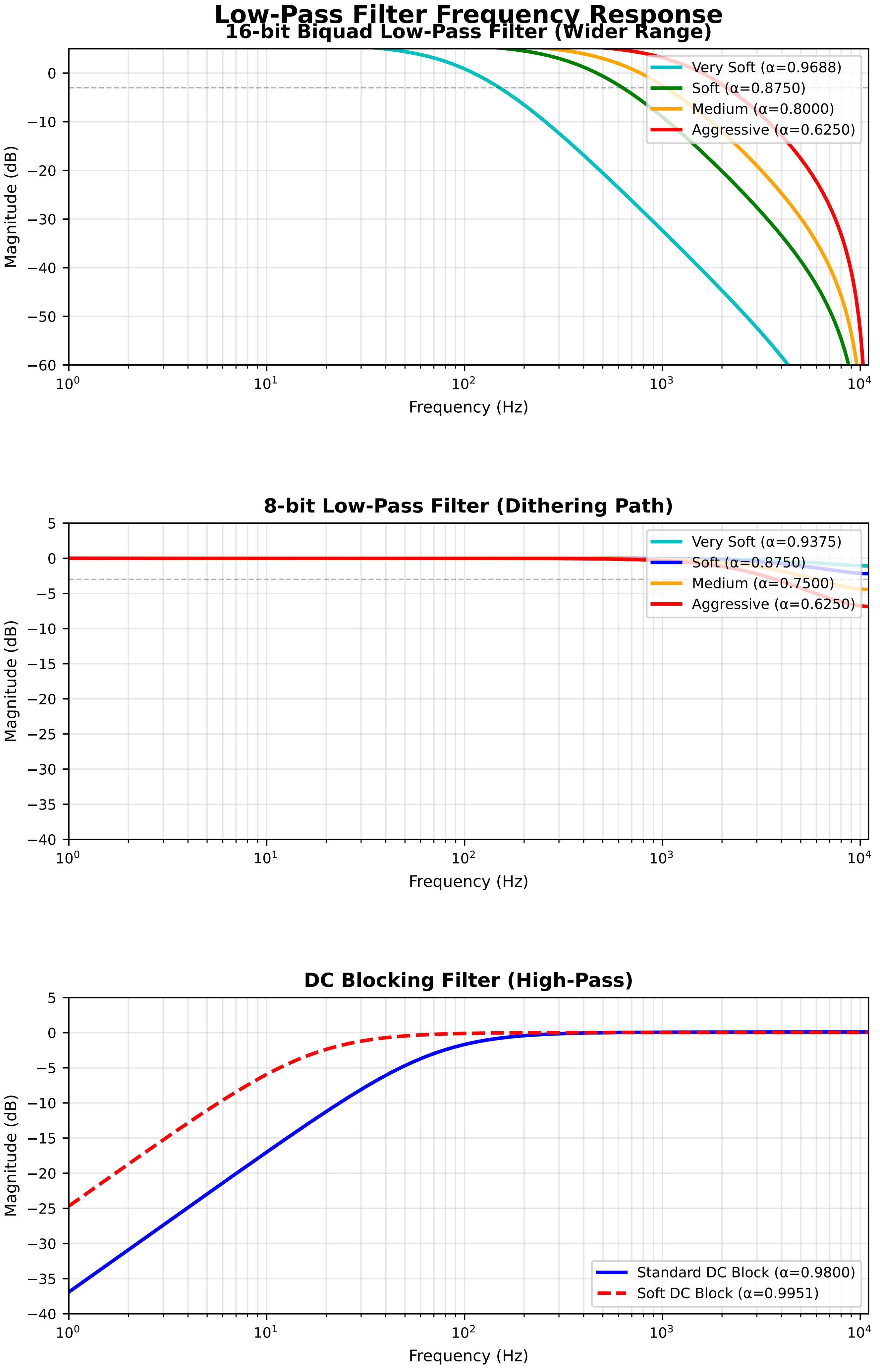
Sample Rate: 22 kHz (default playback speed)
Nyquist: 11 kHz (maximum usable frequency)
Word Size: 16-bit signed integer audio
Architecture: Fixed-point integer DSP (no FPU)

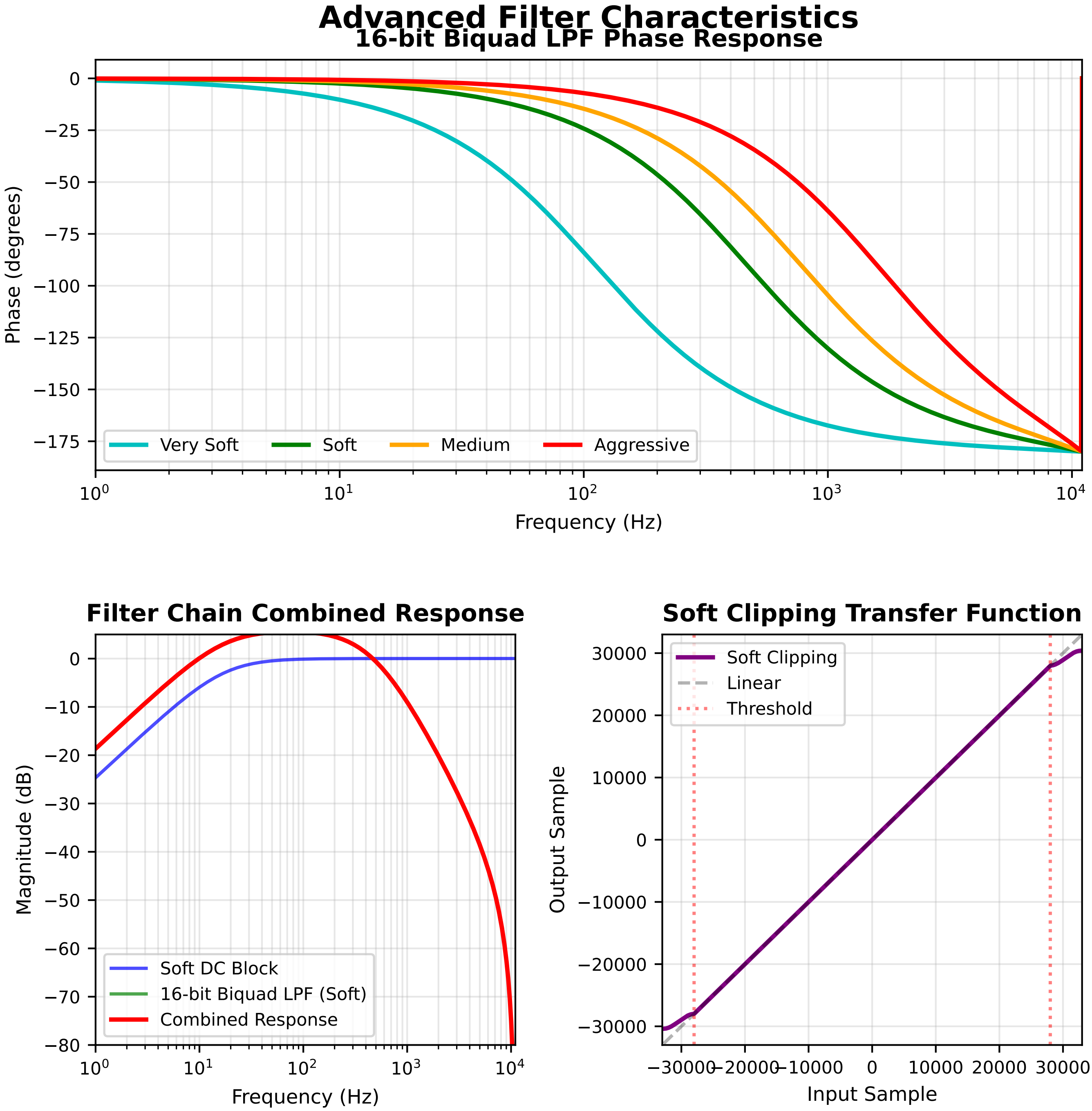
Filter Chain Components:

- 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 (±28,000 threshold, ±85% of full scale)
- Fade in/out effects (configurable)
- Noise gate (optional)
- Dithering for 8-bit samples (TPDF)

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


Low-Pass Filter Frequency Response
16-bit Biquad Low-Pass Filter (Wider Range)
Magnitude (dB)
Frequency (Hz)
Very Soft (α=0.9688)
Soft (α=0.8750)
Medium (α=0.8000)
Aggressive (α=0.6250)
8-bit Low-Pass Filter (Dithering Path)
Magnitude (dB)
Frequency (Hz)
Very Soft (α=0.9375)
Soft (α=0.8750)
Medium (α=0.7500)
Aggressive (α=0.6250)
DC Blocking Filter (High-Pass)
Magnitude (dB)
Frequency (Hz)
Standard DC Block (α=0.9800)
Soft DC Block (α=0.9951)

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

16-bit Biquad LPF:                          8-bit Low-Pass Filter:
  Very Soft:       150 Hz ( 0.68% Fs)    Very Soft:    11000 Hz (50.00% Fs)
  Soft:            630 Hz ( 2.86% Fs)    Soft:         11000 Hz (50.00% Fs)
  Medium:         1042 Hz ( 4.74% Fs)    Medium:        5893 Hz (26.79% Fs)
  Aggressive:     2115 Hz ( 9.61% Fs)    Aggressive:    3734 Hz (16.97% Fs)

Note: Aggressive setting on 16-bit filter uses startup warm-up (8 passes) to eliminate transient cracking.
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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 (α = 0.98): ~44 Hz cutoff @ 22 kHz
- Soft (α = 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: α = 0.875 (cutoff ~6800 Hz)
- Medium: $\alpha \approx$ 0.80 (cutoff ~5900 Hz)
- Aggressive: α = 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: α = 0.9375 (cutoff ~3200 Hz)
- Soft: α = 0.875 (cutoff ~2800 Hz)
- Medium: α = 0.75 (cutoff ~2300 Hz)
- Aggressive: α = 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

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
- Memory Footprint: ~12 KB code + state vars
- Latency: ~50 ms (playback buffer)
- 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 (≈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.