Apex Amp: Advanced Technical & DSP Handbook (v5.6 Sovereign Modular)

01. Advanced Architecture & Audio Thread Integrity

Apex Amp is built on a modular C++ audio engine (`AmpEngine`) using JUCE 8 and custom SIMD-hardened DSP kernels. The engine operates on standard 64-bit double-precision floating-point buffers and employs a lock-free Read-Copy-Update (RCU) topological graph sorter (`SignalGraph`) to allow dynamic, zero-latency module reordering.

Key Engine Specifications:

- **Audio Thread Hardening**: Zero heap allocations (`malloc`/`new`), zero string operations, and zero disk I/O inside `processBlock()`.
- **Multi-Stage Polyphase Oversampling**: Supports 1x (bypass), 2x, 4x, and 8x factors (`oversampleFactor`) with minimum phase-ripple.
- **Denormal Protection**: Scoped denormal guards (`juce::ScopedNoDenormals`) and subnormal flush checks across all filter state variables.
- **Zipper-Noise Prevention**: Linear 20ms parameter ramps (`juce::SmoothedValue`) for all gains, mixes, and continuous coefficients.
- **Thread-Safe IR Loading**: Impulse response files loaded asynchronously on the Message Thread via `ValueTree::Listener` delegates.

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02. The Ten-Layer AI Reasoning Engine (`AiParameterService`)

The built-in AI Assistant communicates with external LLM providers (Gemini 2.0 Flash, LM Studio, etc.) via a non-blocking asynchronous REST client (`NetworkClient`). The AI interprets user intent and outputs JSON parameter maps governed by the **Ten-Layer Reasoning Framework**.

```
graph TD
    UserPrompt[User Request] --> Network[NetworkClient Async Thread]
    Network --> LLM[LLM Engine Gemini 2.0 / LM Studio]
    LLM --> JSON[Raw JSON Response]
    JSON --> Sanitizer[AiParameterService Sanitizer & Safety Bounds]
    Sanitizer --> APVTS[JUCE APVTS Parameter Commit]
    APVTS --> Smoothing[20ms Smoothed Ramps]
    Smoothing --> DSP[Audio Thread Processing]
```

Framework Layer Breakdown:

1. **Layer 0 (Hardware Anchor)**: Evaluates `srcProfile` (Humbucker, Single-Coil, P90, Active, Piezo, Acoustic) to calculate input energy delta compensation.
2. **Layer 1 (Chronological Circuitry)**: Selects distortion engine (`nuclearMode`: 0=Boutique, 1=Hot-Rod, 2=Nuclear) and active tube stages (`clipStages`).
3. **Layer 2 (Gain Staging)**: Balances preamp drive, power amp push, and output rig trim to maintain unity perceived gain.
4. **Layer 3 (Pre-Gain Shaping)**: Applies surgical 12dB/octave HPF (`preEqHpfFreq`) and 200Hz mute tamer notch (`tamerDepth`) prior to non-linear clippers.

5. **Layer 4 (Space Scaling)**: Sets Schroeder-Moorer reverb tank size (`reverbSize`) and pre-delay offsets (`reverbPreDelay`).
6. **Layer 5 (Power-Amp Bloom)**: Models dynamic envelope B+ supply voltage ducking (`paSag`, `paSagAttack`, `paSagRelease`).
7. **Layer 6 (Width Character)**: Controls Mid/Side matrix expansion (`widthSpread`) and Haas micro-delays (`widthHaas`).
8. **Layer 7 (Stereo Calibration)**: Calibrates M/S balance (`msBalance`) and side air high-shelf filtering (`msSideAir`).
9. **Layer 8 (Purity & Inertia)**: Enforces mutual exclusivity rules (e.g., disabling compressor when `paSag > 0.20`).
10. **Layer 9 (Architectural Integrity)**: Clamps all 293 parameters against structural min/max safety limits.

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03. Advanced Parameter Manifest (293 Registered Parameters)

Group 0. System & Topology Mapping

PARAMETER ID	DATA TYPE & RANGE	DEFAULT	DSP ALGORITHM / TRANSFER FUNCTION	THREAD & SAFETY CONSTRAINT
`masterBypass`	`AudioParameterBool`	`false`	Hard bypass logic switch in `processBlock()`.	Thread-safe atomic read.
`modularEnabled`	`AudioParameterBool`	`false`	Toggles RCU topological sorter execution path.	Lock-free graph pointer swap.
`signal_graph`	`AudioParameterString`	`""`	JSON serialized topology blueprint.	Deserialized on Message Thread.
`utility_nodes`	`AudioParameterString`	`"[]"`	JSON serialized utility node matrix.	Deserialized on Message Thread.

Group 1. Dynamics & Pre-Processing

PARAMETER ID	DATA TYPE & RANGE	DEFAULT	DSP ALGORITHM / TRANSFER FUNCTION	THREAD & SAFETY CONSTRAINT
`masterInputGain`	`AudioParameterFloat` (-18 to +18 dB)	`0.0 dB`	Linear gain multiplier: $G = 10^{\text{dB}/20}$.	20ms linear smoothed ramp.
`inputGain`	`AudioParameterFloat` (-18 to +18 dB)	`0.0 dB`	Preset artist input calibration trim.	20ms linear smoothed ramp.
`tunerEnabled`	`AudioParameterBool`	`false`	Autocorrelation pitch tracking (30Hz–1000Hz).	Tapped pre-gate; audio muted.
`gateEnabled`	`AudioParameterBool`	`true`	Surgical noise gate enable.	1ms lookahead RMS detector.
`gateThreshold`	`AudioParameterFloat` (-100 to 0 dB)	`-60 dB`	Envelope detector threshold pivot.	Smoothed envelope follower.
`gateAttack`	`AudioParameterFloat` (0.01 to 100 ms)	`1.0 ms`	Upward attack coefficient: $\alpha = e^{-1/(f_s \cdot \tau)}$.	Clamped $\tau > 0.01\text{ms}$.

PARAMETER ID	DATA TYPE & RANGE	DEFAULT	DSP ALGORITHM / TRANSFER FUNCTION	THREAD & SAFETY CONSTRAINT
`gateRelease`	`AudioParameterFloat` (1.0 to 500 ms)	`50 ms`	Downward release coefficient: $\beta = e^{-1/(f_s \cdot \tau)}$.	Clamped $\tau > 1.0\text{ms}$.
`srcProfile`	`AudioParameterChoice` (0 to 8)	`0`	Hardware anchor profile (Humbucker, Strat, etc.).	Read-only anchor for AI logic.
`targetProfile`	`AudioParameterChoice` (0 to 8)	`0`	SVF resonant peak transformation matrix.	Biquad filter coefficient update.
`profileDepth`	`AudioParameterFloat` (0.0 to 1.0)	`0.0`	Logarithmic interpolation of SVF profile filter.	Smoothed coefficient crossfade.
`cableCapacitance`	`AudioParameterFloat` (0.0 to 1.0)	`0.0`	High-shelf attenuation filter with resonant shift.	1st-order IIR high shelf.
`pitchEnabled`	`AudioParameterBool`	`false`	PS-SOLA pitch-shifter switch.	Circular grain buffer allocation.
`pitchSemitones`	`AudioParameterFloat` (-24 to +12 ST)	`0.0`	Single-voice PS-SOLA pitch shift ($2^{\text{ST}/12}$).	Cross-faded read-head index.
`pitchMix`	`AudioParameterFloat` (0.0 to 1.0)	`0.0`	Parallel-preserving mix law ($G_{\text{dry}} = \min(1.0, 2 - 2x)$).	Prevents drive gain loss.
`compEnabled`	`AudioParameterBool`	`false`	Soft-knee feedback compressor switch.	RMS feedback detector path.
`compThreshold` / `compRatio`	`Float` (-60 to 0 dB / 1:1 to 20:1)	`-20 / 4:1`	Log-domain gain reduction curve calculation.	Non-linear static transfer curve.
`compAttack` / `compRelease` / `compGain`	`Float` (0.1-100ms / 1-500ms / 0-24dB)	`10 / 100 / 0`	RMS-to-peak smoothing timing & makeup gain.	Clamped non-zero timing constants.
`wahEnabled` / `wahSens` / `wahRes` / `wahFreq`	`Bool` / `Float` (0-1 / 0-1 / 20-5000Hz)	`false / 0.5`	Level-responsive TPT State Variable Filter.	Decoupled SVF filter structure.

Group 2. Turbo Synth PRO Workstation (51 Parameters)

PARAMETER ID	DATA TYPE & RANGE	DEFAULT	DSP ALGORITHM / TRANSFER FUNCTION	THREAD & SAFETY CONSTRAINT
`synthEnabled` / `synthMix` / `synthDrive`	`Bool` / `Float` (0-1 / 0-1)	`false / 1.0 / 0.5`	Master enable, crossfade mix, and soft clipper.	20ms smoothed crossfade.
`synthOsc1Wave` / `synthOsc2Wave` / `synthSubWave`	`Choice` (0 to 8 / 0 to 4)	`0`	Poly-BLEP anti-aliased waveform generators.	Band-limited step correction.
`synthOsc1Level` / `synthOsc2Level` / `synthSubLevel`	`Float` (0.0 to 1.0)	`1.0 / 0 / 0`	Parallel oscillator mixing bus gains.	Smoothed linear gain.
`synthOctave` / `synthOsc2Octave` / `synthSubOctave`	`Choice` (-2 to +2 Octaves)	`0`	Frequency multiplier scaling (2^{octave}).	Checked integer bounds.
`synthOsc1Interval` / `synthOsc2Interval`	`Float` (0 to 12 Semitones)	`0.0`	Chromatic semitone frequency offset.	Frequency multiplier.
`synthOsc2Detune`	`AudioParameterFloat` (±100 Cents)	`0.0`	Fine pitch detune offset: $2^{\text{cents}/1200}$.	Float pitch offset calculation.
`synthGlide`	`AudioParameterFloat` (0 to 1000 ms)	`0.0 ms`	Exponential frequency portamento smoothing.	1st-order LPF pitch filter.
`synthCutoff` / `synthResonance`	`Float` (20-20000Hz / 0-0.99)	`2000 / 0.5`	4-Stage Moog ladder LPF with Q feedback.	Stable Q limiter (<0.99).
`synthEnvAttack` / `synthEnvRelease`	`Float` (0.1-1000ms / 1-3000ms)	`10 / 100`	Follower envelope smoothing constants.	Clamped timing limits.
`synthWavefold`	`AudioParameterFloat` (0.0 to 1.0)	`0.0`	Trigonometric wavefolding: $y = \sin(\pi \cdot x \cdot G)$.	Soft-clipped wavefolder.
`synthNoiseLevel`	`AudioParameterFloat` (0.0 to 1.0)	`0.0`	Pseudo-random white noise generator.	PRNG float generator [0,1].
`synthRingMod`	`AudioParameterFloat` (0.0 to 1.0)	`0.0`	Heterodyne ring modulation: $y = x_{\text{guitar}} \cdot x_{\text{osc}}$.	Audio-rate multiplication.
`synthFmDepth`	`AudioParameterFloat` (0.0 to 1.0)	`0.0`	Frequency modulation depth (Osc 1 -> Osc 2).	Phase increment modulation.
`synthGateThreshold`	`AudioParameterFloat` (-100 to 0 dB)	`-40 dB`	Dual-threshold Schmitt trigger gate.	Prevents gate chattering.

Group 3. High-Gain Preamp & Overdrive

PARAMETER ID	DATA TYPE & RANGE	DEFAULT	DSP ALGORITHM / TRANSFER FUNCTION	THREAD & SAFETY CONSTRAINT
`ampBypass` / `preampBypass`	`AudioParameterBool`	`false`	Complete bypass switches for head and preamp.	Direct bypass branching.
`boostEnabled` / `boostGain`	`Bool` / `Float` (0 to +20 dB)	`false / 0dB`	Germanium FET high-shelf treble boost.	1st-order IIR high shelf.
`fuzzEnabled` / `fuzzGain` / `fuzzTone`	`Bool` / `Float` (0-1 / 0-1)	`false / 0.5`	Asymmetric soft-clipping fuzz algorithm.	DC offset removal filter.
`driveType` / `driveGain` / `driveTone`	`Choice` / `Float` (0-1 / 0-1)	`0 / 0.5 / 0.5`	Overdrive models (Green TS, OCD, Klon).	Non-linear diode clipping knee.
`driveMidFreq` / `driveMidGain` / `driveMidQ`	`Float` (400-3000Hz / ±12dB)	`1600 / +6dB`	Parametric mid-boost bell filter.	Biquad bell filter update.
`preampGain`	`AudioParameterFloat` (0.0 to 1.0)	`0.5`	Multi-stage exponential tube saturation drive.	$G = e^{5.0 \cdot x}$.
`clipStages`	`AudioParameterFloat` (1.0 to 8.0)	`4.0`	Dynamic serial triode stage allocation loop.	Fixed loop upper bound (8).
`nuclearMode`	`AudioParameterChoice` (0, 1, 2)	`0`	Saturation topology (Boutique, Hot-Rod, Nuclear).	Bias floor configuration switch.
`biasAmount`	`AudioParameterFloat` (0.0 to 1.0)	`0.5`	DC bias offset injection into soft clippers.	High-pass DC blocker post-clipper.
`interStageLpf`	`AudioParameterFloat` (1k to 20kHz)	`16000 Hz`	1st-order inter-stage LPF damping filter.	Clamped $f_c \le f_s/2$.
`tamerDepth`	`AudioParameterFloat` (0.0 to 1.0)	`0.0`	Sidechain 200Hz band-reject mute tamer filter.	Dynamic notch gain adjustment.
`preEqHpfFreq`	`AudioParameterFloat` (20 to 500 Hz)	`80 Hz`	Pre-gain 12dB/octave high-pass filter.	2nd-order Butterworth HPF.

Group 4. Power Amp & Studio Cabinet Workstation

PARAMETER ID	DATA TYPE & RANGE	DEFAULT	DSP ALGORITHM / TRANSFER FUNCTION	THREAD & SAFETY CONSTRAINT
`tsBass` / `tsMid` / `tsTreble` / `tsPresence`	`AudioParameterFloat` (0.0 to 1.0)	`0.5`	Passive 3-band Fender/Marshall tone stack EQ.	Bilinear transform biquads.
`powerAmpBypass`	`AudioParameterBool`	`false`	Power amp simulator bypass switch.	Audio thread bypass branch.
`paGain`	`AudioParameterFloat` (0.0 to 2.0)	`0.5`	Output transformer drive scaling.	Linear gain scaling.
`paSag`	`AudioParameterFloat` (0.0 to 1.0)	`0.2`	Dynamic B+ supply voltage envelope compression.	Attack/Release envelope follower.
`paSagAttack` / `paSagRelease`	`Float` (0.1-50ms / 1-1000ms)	`2ms / 100ms`	Voltage drop attack and bloom recovery timing.	Clamped non-zero timing.
`paPresenceGain` / `paDepthGain`	`Float` (-15 to +15 dB)	`0 dB`	Power amp top-end presence & low-end depth.	High shelf & Low shelf filters.
`cabType`	`AudioParameterChoice` (0 to 30)	`1`	FFT convolution impulse response selection.	Zero-latency uniform-partitioned FFT.
`cabTilt` / `cabDistance`	`Float` (0.0 to 1.0)	`0.2 / 0.2`	Off-axis tilt EQ (650Hz pivot) & air proximity.	Serial tilt filter & 1-15ms delay.
`cabModernShaping`	`AudioParameterBool`	`false`	Dual notch filter removing speaker fizz (350Hz/5.8kHz).	Dual band-reject notch filters.
`dualMicEnabled`	`AudioParameterBool`	`false`	Dual mic parallel convolution studio toggle.	Parallel convolution streams.
`micAType` / `micBType`	`Choice` (0 to 5)	`0`	Mic character profiles (SM57, MD421, R121, etc.).	Parametric mic coloring filter.
`micATilt` / `micADist` / `micALevel`	`Float` (0-1 / 0-1 / 0-2)	`0.35 / 0.15`	Mic A position, proximity distance, and level.	Parametric positioning filters.
`dualMicBlend`	`AudioParameterFloat` (0.0 to 1.0)	`0.4`	Equal-power linear mic crossfader ($A \cdot (1-x) + B \cdot x$).	Smoothed crossfade multiplier.

Group 5. Post-FX, Ambience & Master Suite

PARAMETER ID	DATA TYPE & RANGE	DEFAULT	DSP ALGORITHM / TRANSFER FUNCTION	THREAD & SAFETY CONSTRAINT
`chorusEnabled` / `chorusRate` / `chorusMix`	`Bool` / `Float` (0.1-10Hz / 0-1)	`false / 0.5`	Multi-voice delay LFO sweep with stereo phase.	3rd-order Lagrange interpolation.
`flangerEnabled` / `flangerRate` / `flangerFeedback`	`Bool` / `Float` (0.05-5Hz / 0-0.95)	`false / 0.5`	Resonant comb-filtering flanger algorithm.	Feedback saturation clamp.
`phaserEnabled` / `phaserRate` / `phaserDepth`	`Bool` / `Float` (0.1-10Hz / 0-1)	`false / 0.5`	Multi-stage all-pass phase sweeping filter.	Cascade all-pass filters.
`tremoloEnabled` / `tremoloRate` / `tremoloDepth`	`Bool` / `Float` (0.5-20Hz / 0-1)	`false / 0.5`	Amplitude modulation tremolo with soft LFO.	Multiply amplitude envelope.
`postEqEnabled` + 10 Band Gains	`Bool` / `Float` (-15 to +15 dB)	`false / 0 dB`	ISO 10-band graphic equalizer (31Hz to 16kHz).	Parallel/Serial biquad bandbank.
`widenerEnabled` / `widthSpread` / `widthHaas`	`Bool` / `Float` (0-1 / 0-30ms)	`false / 0 / 0`	Mid/Side matrix expansion & Haas micro-delay.	Mid/Side linear decoding matrix.
`msBalance` / `msSideAir`	`Float` (-1 to +1 / 0 to 1)	`0.0 / 0.0`	M/S focus balance & 10kHz side air high-shelf.	Side channel high shelf boost.
`delayEnabled` / `delayTime` / `delayFeedback`	`Bool` / `Float` (1-2000ms / 0-1)	`false / 350 / 0.4`	Elite stereo delay 1 circular buffer engine.	Hermite fractional delay reader.
`delayShimmer` / `delayDucking`	`Float` (0.0 to 1.0)	`0.0 / 0.0`	Granular pitch-shifter feedback & sidechain ducking.	Ducking envelope follower.
`reverbEnabled` / `reverbMix` / `reverbSize`	`Bool` / `Float` (0-1 / 0-1)	`false / 0.2`	Schroeder-Moorer tank (Spring, Plate, Hall, etc.).	Parallel comb & all-pass filters.
`reverbShimmer` / `reverbDucking`	`Float` (0.0 to 1.0)	`0.0 / 0.0`	Octave pitch shifter in tank feedback & ducking.	Soft-clipping feedback saturator.
`masterVol`	`AudioParameterFloat` (0.0 to 1.0)	`0.8`	Smoothed master rig output trim.	20ms linear smoothed precision gain.

PARAMETER ID	DATA TYPE & RANGE	DEFAULT	DSP ALGORITHM / TRANSFER FUNCTION	THREAD & SAFETY CONSTRAINT
`masterLimiter`	`AudioParameterBool`	`true`	Peak lookahead brickwall safety limiter (0dBFS).	Lookahead circular peak buffer.

Group 6. Universal CV Modulation Matrix (28 Parameters)

PARAMETER ID	DATA TYPE & RANGE	DEFAULT	DSP ALGORITHM / TRANSFER FUNCTION	THREAD & SAFETY CONSTRAINT
`modLfo1Rate` / `modLfo1Wave`	`Float` (0.05-30Hz) / `Choice`	`1.0 Hz / Sine`	Multi-waveform CV LFO generator.	Clamped rate bounds.
`modLfo2Rate` / `modLfo2Wave`	`Float` (0.05-30Hz) / `Choice`	`1.0 Hz / Sine`	Multi-waveform CV LFO generator.	Clamped rate bounds.
`modEnvThresh` / `modEnvAttack` / `modEnvRelease`	`Float` (-100-0dB / 0.1-1000ms)	`-30dB / 10ms`	Envelope follower tracking peak audio amplitude.	Smoothed envelope signal.
`modAdsrAttack` / `modAdsrDecay` / `modAdsrSustain` / `modAdsrRelease`	`Float` (0.1-1000ms / 1-3000ms)	`10ms / 100ms`	4-Stage transient ADSR generator triggered by picks.	Non-zero timing constants.
`modSlot1...8Source` / `modSlot1...8Dest` / `modSlot1...8Depth`	`Choice` / `Choice` / `Float` (-1 to +1)	`None / None / 0`	8 Visual CV matrix slots routing modulators to 150+ APVTS targets.	Clamped bipolar depth.

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04. Real-Time Hardening & Safety Verification

1. **Zero Heap Allocation Rule**: Pre-allocate all circular delay buffers, convolution partitions, and state-variable filters during `prepareToPlay()`.
2. **Scoped Denormal Guards**: Enforce `juce::ScopedNoDenormals` at the entry of `AmpEngine::process()`.
3. **Feedback Explosion Shield**: Dynamic `std::tanh()` saturation integrated into delay and reverb feedback loops.
4. **Thread Isolation**: Communication between Message Thread and Audio Thread utilizes lock-free atomic primitives and ValueTree listeners.
