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1. 产品概览

LiVocal 将前级、动态、音色和自适应处理放在一块清晰的面板上。先选基础音色，再调整明暗、厚度与动态。需要时独立开关模块，或改变它们的顺序。

金色、银色、珍珠与黄铜：四种独特的音乐风格音色曲线，给你四个不同的起点。

本手册以 1.0.5 新建实例为基准：金色音色、标准音质、实体界面、100% 界面尺寸；各声音模块启用，音色与空气感、厚度为 0%，混合为 100%。

使用位置

- 用于单声道或双声道人声轨。输入和输出声道数匹配；本产品没有外部侧链输入。
- 先完成必要的剪辑、底噪处理和大幅片段音量修正，再用 LiVocal 塑形与控制。具体处理顺序由音源和目标决定。

说明

LiVocal 不提供音高修正、节奏编辑或自动配唱。AIR、BODY、TONE 为 0% 也不等于整条处理链旁路：其他已开启模块及当前基础音色仍然参与处理。

阅读路径

需要完成	阅读位置
第一次使用	安装与授权、快速开始
查找控件	界面地图、参数参考
比较或保存声音	工具栏与预设
选择录音或导出设置	音质、延迟与工程恢复
排查无声或异常	故障排查

2. 安装与授权

项目	安装包与宿主
产品版本	LiVocal 1.0.5
系统要求	以对应平台当版官方下载页与安装包为准；宿主可能有更高要求。
插件格式	AU / VST3 / AAX Native
声道	单声道、双声道；无外部侧链。

安装完整官方安装包，再在宿主中使用对应格式：Logic 使用 AU，Pro Tools 使用 AAX，Studio One 与 REAPER 可按平台使用 VST3。插件版本与共享授权管理器版本分别显示。

1. 从 JMECHO 官方渠道取得与系统及宿主匹配的安装包，运行安装器并完成系统授权。
2. 安装完成后打开 JMECHO 授权管理系统。安装器可能自动打开；未自动打开时手动启动。
3. 按管理器提示激活付费许可，或在试用开放时启动试用；在插件的“授权信息”中确认状态。
4. 重新打开 DAW，必要时重新扫描对应格式，在音频轨上插入 LiVocal。
5. 播放人声，确认输入、输出表有响应，再保存工程。

付费许可与离线工作

付费许可为永久许可，不是订阅。每份许可提供 2 个设备授权槽位，并允许在任意滚动 365 日内最多 3 次自助设备迁移。首次付费激活与迁移需要通过独立授权辅助程序联网；有效永久许可可保存在本机后，可在其覆盖设备上离线使用。

系统重装、删除设备密钥或授权文件可能需要重新验证。设备遗失、迁移额度或恢复问题请联系支持；不要自行删除授权资料来尝试“重置”。

七日试用

试用从服务器首次成功签发起连续计算 7×24 小时，不自动购买或扣款。试用启动、每次新的宿主进程启动及短期许可刷新时需要联网；单次短期许可最长 24 小时，且不超过试用结束时间。

注意

没有有效付费许可或有效试用短期许可时，LiVocal 不能输出音频。开始重要录音、离线工作或导出前，请先确认授权状态。独立辅助程序需要自身获准联网。

3. 快速开始

选择一段包含轻声、正常发声和较强音节的人声循环。下面的顺序是一种起点；以实际录音的变化和听感决定最终设置。

1. 以新建实例开始，先保持金色音色、标准音质和混合 100%。播放时确认 RAW 和 OUT 表有信号。
2. 先听前级：分别比较染色和压缩开关。需要保留较低的人声基频或录音已经低切时，比较低切开关的影响。
3. 调整校准而非把它当作“压缩量”。50% 为零校准增益；高于或低于中心改变送入处理链的电平及处理程度。
4. 比较四种基础音色，再少量调整音色旋钮。0% 保留所选基础音色；负值偏暗，正值偏亮。
5. 按需要调整空气感、厚度和靠前。再用音色稳定、去齿音和柔化控制不均匀或突出的成分。
6. 如需并行混合，降低 MIX。将指针移到 OUTPUT 旋钮，点击出现的 AUTO，并继续播放有效人声直到测量完成。
7. AUTO 完成后比较处理前后，并按听感微调 OUTPUT。需要复用时，在 DAW 的插件预设栏保存设置，同时保存工程。

说明

AUTO 以约 2 秒有效音频比较输入和输出，并一次性写入 OUTPUT。它不是持续跟随器，也不是设定绝对 LUFS 的工具。

建议

先听变化，再决定是否保留。默认开启的处理不是适合所有音源的固定答案；可逐个关闭模块，确认每一项处理带来的实际收益。

4. 界面地图



图 1. LiVocal 1.0.5 实体界面，新建实例设置。下方编号定位对应模块，具体控件见后续局部图；读数示意图状态。

区域	查阅编号	功能
前级 / PRE	1-4	低切、染色、前级压缩
动态控制 / DYNAMICS	5-7	动态模块、全局校准与稳幅
音色塑形 / CHARACTER	8-11	四种基础音色、音色倾斜与温暖开关
润色 / POLISH	12-15	空气感、厚度与靠前
自适应 / ADAPTIVE	16-19	均衡范围、去齿音与柔化
总控 / MASTER	20-23	混合、输出、AUTO 与全局开关

基本操作与反馈

上下拖动旋钮：向上增大，向下减小。按住 Command 或 Shift 时以较小幅度调节；拖动中按下或松开修饰键不会跳值。点击数值框可输入数值，回车确认，Esc 取消。手动百分比按 1% 步进，dB 按 0.1 dB 步进；宿主自动化仍保留参数本身的小数精度。

模块标题旁的开关控制该模块。拖动标题左侧的三横线把五个声音模块重新排序；总控始终位于末端。RAW 为原始输入，CAL 为经校准和内部输入平衡后的电平，OUT 为最终输出。显示图用于观察处理活动，不是独立的可拖动频段编辑器。

说明

简洁界面与实体界面使用同一套声音参数。MID 是中频存在感开关；EQ RANGE 为自适应均衡允许的最大校正范围。切换界面不改变当前声音。

5. 参数参考

编号沿用界面图。下列默认值指新建 1.0.5 实例；恢复工程、加载预设或切换 A/B 后可能不同。关掉模块会旁路其声音处理，但校准与稳幅属于独立控制，详见相应条目。

前级



图 2. 前级控件 1-4。低切先于前级染色和压缩；图中开关均启用。

1. 前级开关 / PRE ON

启用或旁路整个前级模块。

范围: 开 / 关 / 默认: 开 / 单位: 不适用

变化: 开启后，低切、染色和压缩按各自设置处理。

关联: 全局电源需开启；关闭前级不关闭其他模块。

2. 低切 / LOW CUT (CUT)

在前级最前端加入固定高通滤波。

范围: 开 / 关；90 Hz，12 dB/oct / 默认: 开 / 单位: 开关

变化: 减少低频隆隆声及低频能量；也会影响低音人声的低频成分。

关联: 仅在前级启用时生效。

提示: 录音已低切时，比较开关前后的厚度。

3. 染色 / COLOR

启用前级染色部分。

范围: 开 / 关 / 默认: 开 / 单位: 不适用

变化: 改变前级的谐波与声音质感，不同时打开压缩。

关联: 需要前级开启；与压缩开关独立。

4. 压缩 / COMP

启用前级压缩及关联的动态共振抑制。

范围: 开 / 关 / 默认: 开 / 单位: 不适用

变化: 控制前级动态及特定中频突出成分，不同时打开染色。

关联: 需要前级开启；包含固定 900 Hz / 3 kHz 动态处理。

提示: 先比较压缩，再比较染色，可更清楚辨认各自贡献。

动态控制



图 3. 动态区域 5-7。6 是全链输入校准，7 是模块后独立稳幅控制。

5. 动态模块开关

启用或旁路动态控制模块。

范围: 开 / 关 / 默认: 开 / 单位: 不适用

变化: 开启多频段动态控制、关联的人声自适应处理及内部输入平衡。

关联: 关闭后，校准和稳幅仍可独立操作。

6. 校准 / CALIBRATE

调整进入声音处理链的电平，并在链后施加反向校准增益。

范围: 0-100% / 默认: 50% / 单位: %

变化: 0%、50%、100% 分别对应 -12、0、+12 dB 输入校准。提高校准可改变压缩和非线性处理程度；反向补偿不抵消这些声音变化。

关联: 属于全局输入校准，动态模块关闭时仍有效。

提示: 从 50% 起调，并同时观察 CAL 表和聆听。

注意: 它不是压缩百分比，也不是绝对响度目标。

7. 稳幅 / STABILIZER (STAB)

对前级和动态处理引起的动态电平损失进行跟踪补偿。

范围: 开 / 关 / 默认: 开 / 单位: 不适用

变化: 补偿在可移动模块之后施加一次，避免把音色塑形的变化直接当成压缩损失。

关联: 独立开关；不是 OUTPUT AUTO，也不把输出持续锁定到某个 LUFS 值。

音色塑形



图 4. 音色区域 8-11。音色 0% 保留所选基础音色，不旁路基础响应。

8. 音色模块开关

启用或旁路基础音色与音色倾斜处理。

范围: 开 / 关 / 默认: 开 / 单位: 不适用

变化: 关闭后，四种音色、温暖和音色旋钮不再参与该模块处理。

关联: 不关闭前级、润色或智能模块。

9. 基础音色选择

选择一组固定基础音色响应。

范围: 金色、银色、珍珠、黄铜 / 默认: 金色 / 单位: 选项

变化: 四种响应可在同一段音源上比较。不存在适用于所有歌手的固定选择。

关联: 需要音色模块开启; 与温暖开关共同决定基础响应。

提示: 用 A/B 保存两种候选, 并在相近输出电平下比较。

10. 音色 / TONE

在当前基础音色之后改变明暗倾向。

范围: -100% 至 +100% / 默认: 0% / 单位: %

变化: 负值偏暗, 正值偏亮; 0% 为此倾斜处理的中性位置。两端来自实测响应, 端点不是严格对称的标准搁架。

关联: 新建实例使用实测倾斜均衡, 独立于四种基础音色; 仅在音色模块开启时生效。

提示: 少量调整后再检查齿音、厚度和混音中的位置。

注意: 本项不额外加入谐波或噪声; 整条插件链中的其他部分仍可能有非线性处理。

11. 温暖 / WARM

切换所选基础音色的定义方式。

范围: 开 / 关 / 默认: 开 / 单位: 不适用

变化: 开启温暖定义; 关闭使用原始定义。它是声音选择, 不是只更换面板颜色。

关联: 需要音色模块开启; 切换后重新比较输出电平与听感。

润色

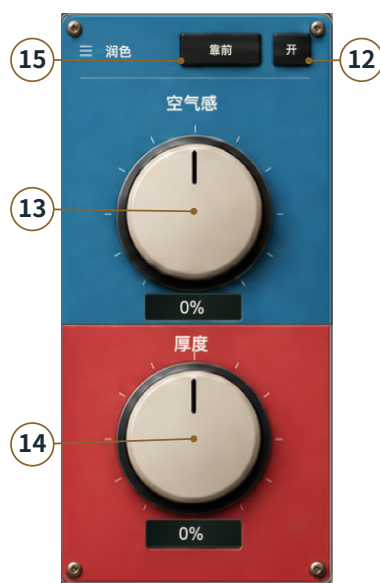


图 5. 润色区域 12-15。空气感与厚度各自有正、负方向。

12. 润色模块开关

启用或旁路润色模块。

范围: 开 / 关 / 默认: 开 / 单位: 不适用

变化: 同时决定空气感、厚度和靠前处理是否参与声音链。

关联: 三个控制仍保存自己的设置。

13. 空气感 / AIR

增加或减少高频空气感。

范围: -100% 至 +100% / 默认: 0% / 单位: %

变化: 正值结合高频均衡与谐波润色；负值作高频削减，0% 为本控制中性。

关联: 需要润色模块开启；去齿音仍是独立动态控制。

提示: 与去齿音交替检查，避免只依据更亮的瞬间判断。

14. 厚度 / BODY

增加或减少声音的低频与厚度成分。

范围: -100% 至 +100% / 默认: 0% / 单位: %

变化: 正值增加厚度并可加入关联润色；负值减少相应成分。

关联: 需要润色模块开启；前级低切也会影响低频结果。

提示: 在混音中检查拥挤程度，而不只听独奏。

15. 中频存在感 / MID

启用固定中频存在感均衡。

范围: 开 / 关; 2000 Hz / 默认: 开 / 单位: 开关

变化: 提升中频存在感, 不额外添加谐波。

关联: 需要润色模块开启。

提示: 如果元音突出或靠前过度, 先比较此开关。

智能处理



图 6. 智能区域 16-19。图中的 dB 数值表示允许的处理范围或上限, 不是当前持续衰减量。

16. 智能模块开关

启用或旁路智能处理模块。

范围: 开 / 关 / 默认: 开 / 单位: 不适用

变化: 控制音色稳定、去齿音和柔化处理是否进入声音链。

关联: 不关闭其他动态或音色模块。

17. 均衡范围 / EQ RANGE

设置自适应音色校正的允许范围。

范围: 0-8 dB / 默认: 3.0 dB / 单位: dB

变化: 较大值允许更明显的校正; 0 dB 关闭此音色校正部分, 不同时关闭去齿音或柔化。

关联: 需要智能模块开启; 新建实例采用可归零的范围。

提示: 逐步增加, 以元音是否自然及词句间音色是否更一致判断。

18. 去齿音 / DE-ESS

设置动态高频平滑的最大衰减。

范围: 0-8 dB / 默认: 3.0 dB / 单位: dB

变化: 遇到突出的高频成分时允许衰减; 0 dB 关闭本项。它不持续固定削减设定的 dB 数。

关联: 需要自适应模块开启; 简洁与实体界面对应同一个去齿音参数。

提示: 先检查含 s、sh 等较明显高频辅音的句子, 再比较整段明亮度。

19. 柔化 / SOFT

启用存在感区域的动态柔化。

范围: 开 / 关 / 默认: 开 / 单位: 不适用

变化: 减轻突出的存在感瞬态; 与去齿音处理范围不同。

关联: 需要智能模块开启; 当前 SOFT 不作为宿主自动化参数。

总控



图 7. 总控区域 20-23。22 指向 OUTPUT 的 AUTO 悬停区域, 按钮在悬停或测量状态下出现; 23 位于顶栏。

20. 混合 / MIX

设置干声与处理后声音的混合比例。

范围: 0-100% / 默认: 100% / 单位: %

变化: 0% 以干声为基础, 100% 为全处理声, 中间值用于并行混合。插件内部对干湿时序及混合电平作相应处理。

关联: OUTPUT 仍控制最终电平; MIX 0% 不等于全局旁路。

21. 输出 / OUTPUT

调整插件最终输出增益。

范围: -18 至 +18 dB / 默认: 0.0 dB / 单位: dB

变化: 增大或减小最终音量; AUTO 会一次性修改此值。

关联: 在干湿混合后作用; 无绝对 LUFS 锁定功能。

提示: 与旁路比较时先校正电平差, 并检查下游是否削波。

22. 自动输出匹配 / AUTO

短时测量输入与输出, 并一次性调整 OUTPUT。

范围: 单次测量; 结果受 OUTPUT 范围限制 / 默认: 未测量 / 单位: 测量动作

变化: 约 2 秒有效音频后, 根据输入与输出的比较结果一次性调整输出增益。无有效信号时最多约 15 秒后结束。

关联: 需有效授权、全局开启并持续播放; 测量时保持设置和自动化稳定。

提示: 出现 DONE 表示已应用; LIMIT 表示达到输出边界; NO SIG 表示没有取得有效测量。

注意: 它只测当前片段。改变音色、顺序、MIX 或音源后, 应重新判断是否需要测量。

23. 全局电源

在授权有效时切换整条插件的处理与旁路。

范围: 开 / 关 / 默认: 开 / 单位: 不适用

变化: 关闭时使用与当前报告延迟匹配的干声。切换采用平滑过渡。

关联: 旁路仍维持内部处理状态, 不等同卸载插件或降低 CPU; 授权规则不因旁路失效。

6. 比较、预设与更多选项

控件	操作与结果
A / B	切换两个声音槽，各自保存声音参数和模块顺序。
COPY	把当前声音复制到另一个槽。
← / →	撤销或重做声音编辑；不可用时仍保持相同文字颜色。
EN / 中	显示点击后将切换的语言。
≡	按音质与导出、界面、帮助与信息分组。

从 DAW 调用预设

插件的更多菜单不再管理预设。使用 DAW 自己的插件预设栏选择工厂预设、保存你的设置，并将工程一起保存。预设参数共用同一份数据，AU、VST3 与 AAX 分别使用对应格式。

名称	起点
01 Gold Default	金色的均衡起点，适合先听再调。
02 Pearl Gentle	温和塑形，保留自然的字句变化。
03 Silver Air	增加高频延展，适合偏暗的录音。
04 Brass Body	增加厚度，适合偏薄的录音。
05 Gold Forward	突出中频存在感，方便在伴奏中定位。
06 Pearl Open	较开放的珍珠音色，轻度动态校正。
07 Warm Close	偏暖、贴近的音色，适合近距离人声。
08 Bright Control	保留明亮度，同时收敛突出的高频。
09 Sibilance Soft	以去齿音为重点，试听后按录音减少或增加处理。
10 Brass Parallel	70% 处理声混合，保留较多原始表现。
11 Clean Shape	关闭前级染色、压缩与动态、自适应模块，以均衡塑形为主。
12 Dry Detail	80% 处理声混合，兼顾细节与原始表现。

试听建议

预设是起点，不会分析录音并自动保证合适。先在同一乐句试听，检查校准、齿音和输出，再按音源调整。所有工厂预设的输出增益为 0 dB。

更多选项

分组	内容
音质与导出	选择低延迟、标准或高音质；设置离线导出采用高音质。
界面	直接切换简洁／实体模式，选择界面尺寸。
帮助与信息	入门说明、用户手册、插件详情、授权信息。

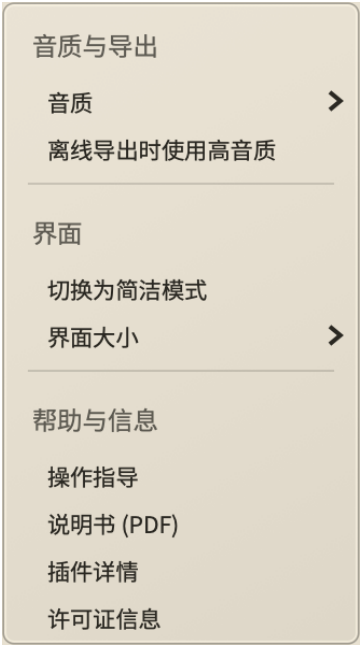


图 8. 更多选项的分组。工厂预设从 DAW 调用。



图 9. 简洁界面。声音控制与实体模式对应。

声音预设与 A/B 比较保留当前界面、语言、音质和离线导出偏好。DAW 保存整个工程状态时也可恢复这些偏好；各宿主自身的预设保存规则可能不同。

7. 音质、延迟与工程恢复

菜单音质	使用方式
低延迟 / LOW LATENCY	用于优先考虑处理开销和较低算法延迟的情形；外部报告延迟仍受离线高品质预留等设置影响。
标准 / STANDARD	新实例默认。适合作为常规处理与声音比较的起点。
高品质 / HIGH QUALITY	更高处理品质与开销；使用前确认系统有足够余量。

插件向宿主报告延迟，并对干湿路径及旁路路径作对齐。不要把质量名称换算成固定的录音总延迟；实际值与采样率、模式、离线高品质预留及宿主有关。当前值可在插件详情和宿主中查看。

录音监听

需要较低延迟时选择低延迟模式，并确认离线高品质偏好关闭。音质模式使用不同的处理方式，不应把它们理解为声音完全相同的速度档位。实际录音总延迟还包含宿主缓冲、音频接口与其他插件。

高品质离线导出

此偏好默认关闭。开启后，宿主以非实时模式调用插件时使用高品质；实时播放或实时导出仍使用手动选择的质量。开启该偏好会为高品质延迟预留并通知宿主。

注意

切换相关设置后，先停止播放并确认宿主延迟更新。若菜单提示等待宿主延迟同步，请先让宿主重新准备或重新加载插件，再离线导出。名称中有“离线”不代表每种导出方式都会触发非实时模式。

恢复既有工程

既有工程可保留其原来的音色控制算法和音色稳定范围。当前界面会据工程状态绑定对应控制，因此恢复工程时看到的范围可能不同于新建实例。只有选择菜单里的新版音色或新版音色稳定升级动作才改变这些声音解释。

注意

升级前保存工程副本并记录自动化。升级不会把原来的自动化轨道自动转成新版语义；尤其音色稳定的新、旧范围使用不同参数。比较音频后再决定采用。

8. 频率、电平与声音响应

先区分单位：Hz、kHz 表示频率；dB 表示两个电平之间的比值；dBFS 表示相对于数字满刻度的信号电平。百分比是控件位置，不是压缩比、频率或持续衰减量。

TONE：明暗倾斜与相位

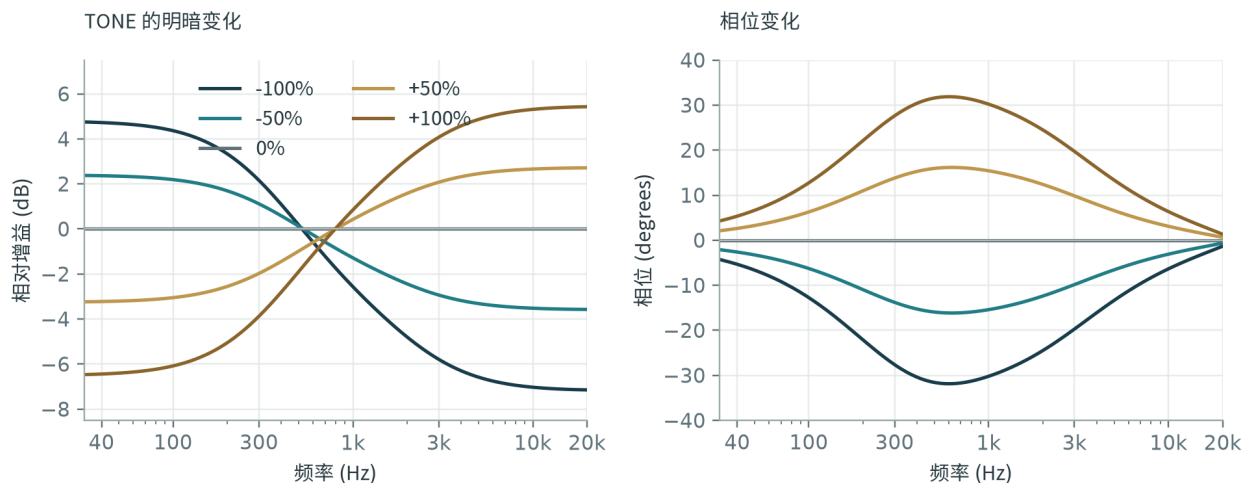


图 10. 当前 TONE 处理单元在 48 kHz 下的实测幅度与相位。测量使用峰值 0.1 的稳定正弦，逐频等待稳定；不包含四种基础音色、前级或其他模块。

0% 在这个处理单元中为中性。负值同时增加低频相对份量、减少高频相对份量；正值向较亮方向倾斜。最大与最小端点分别来自自己选定的实测响应，所以增益并非严格镜像。中间位置渐变到相应端点。

相位曲线表示同一频率的输出波形相对于输入的角度变化。它不等于插件向宿主报告的延迟，也不能据此直接推断整个插件的录音延迟。TONE 本身不增加谐波或噪声，其他已启用模块仍可能改变谐波和动态。

MID：宽中频存在感

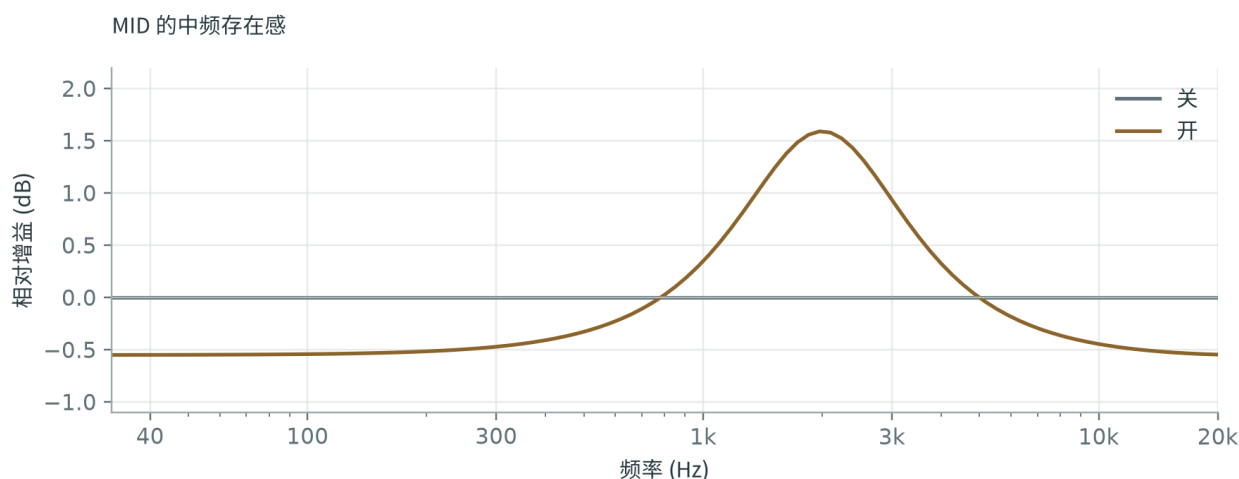


图 11. 48 kHz 下 MID 单元的实测稳态响应。关闭时为中性；开启时为宽中频提升及固定电平补偿。

中心频率为 2 kHz，Q 为 0.78。内部钟形提升为 +2.142 dB，并叠加 -0.55034 dB 电平补偿，因此单元中心的净提升约为 +1.59 dB。Q 描述曲线宽度：这里是宽范围塑形，不能把它当作窄共振消除工具。

MID 在润色模块中作用。它有平滑切换，不是新增的前瞻处理。开启后应在伴奏中比较字句清晰度与元音突出程度。

低切：频率与斜率

前级低切为 90 Hz、12 dB/oct 的固定高通，Q 约为 0.707。90 Hz 附近并不是突然消失的边界；越向低频，衰减越多。12 dB/oct 是远离转折处的渐近斜率，不代表每一个低频点都固定降低 12 dB。

9. 校准与动态处理的含义

CALIBRATE：先改变驱动电平

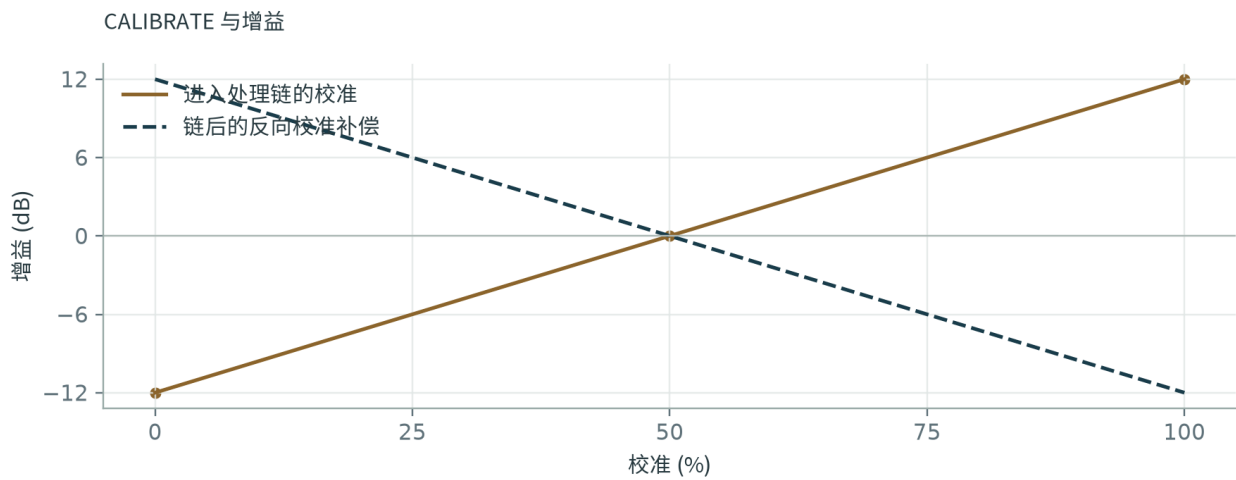


图 12. 校准增益的精确映射；实线为输入校准，虚线为处理链后的反向校准。它们不是整条链的最终响度曲线。

校准位置	输入校准	链后反向校准
0%	-12 dB	+12 dB
25%	-6 dB	+6 dB
50%	0 dB	0 dB
75%	+6 dB	-6 dB
100%	+12 dB	-12 dB

输入校准的关系为 $0.24 \times \text{控件百分比} - 12 \text{ dB}$ ，调节采用约 25 ms 的增益平滑。反向增益只补偿这一步的线性增益；压缩、染色和自适应处理已经受到输入电平影响，因此最终声音不会被“抵消回原样”。它也不取代独立的稳幅或 OUTPUT AUTO。

动态控制：四个检测分区

动态检测的四个分区

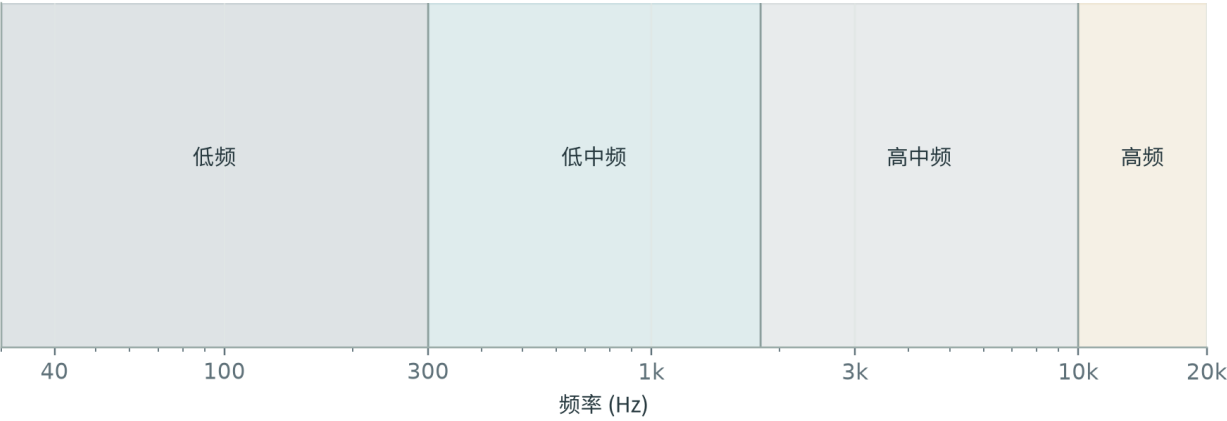


图 13. 动态检测分区示意，分界为 300 Hz、1.8 kHz、10 kHz。色块只说明检测区域，不表示分频点处的增益或砖墙式音频切分。

检测区域	名义启动	基础释放
低频	18 ms	160 ms
低中频	10 ms	110 ms
高中频	5 ms	80 ms
高频	2.5 ms	55 ms

这是当前动态模块内部的名义包络设置，不是可独立调整的四组旋钮。内部基础压缩比为 4:1，并结合宽拐点、参考电平校准、强度映射和快慢包络。持续音与短促音可能产生不同释放行为，表中的时间不能直接当作完整输出曲线到达某个百分比的实测时间。

启动时间描述检测器或增益控制响应的速度，不是为了等待未来音频而加入的延迟。判断处理是否合适，应同时听句子起伏、辅音和呼吸，不只看衰减条。

10. 自适应、去齿音与显示

EQ RANGE：允许变化，不是固定均衡

自适应均衡使用 12 个检测中心：90、140、220、350、550、850 Hz，以及 1.3、2、3、4.5、6.8、10.5 kHz。它比较邻近频率的能量并结合预定音色目标，再平滑更新校正。检测中心不等于 12 个用户可编辑的固定均衡频段。

新建实例的 EQ RANGE 与 DE-ESS 都为 0–8 dB，因此两者在 3.0 dB 时有相同的旋钮角度。EQ RANGE 为 0 时，均衡目标平滑回到中性；DE-ESS 与 SOFT 是其他控制，不会因此全部关闭。把 EQ RANGE 设为 3 dB，不代表整条链每个频率持续提升或削减 3 dB。

DE-ESS：上限与实际衰减

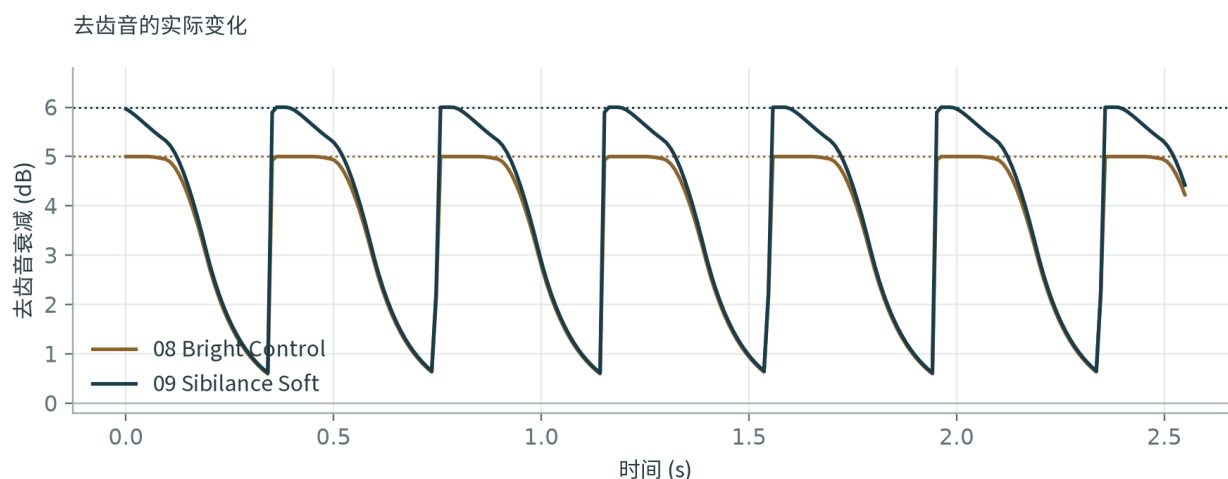


图 14. 两份工厂预设的实际去齿音读数：08 上限 5 dB，09 上限 6 dB。48 kHz、512 样本块，以 300 Hz 基底叠加间歇 8.2 kHz 突发；先等待稳定，每块末记录一次。虚线为设置上限，实线为实际控制衰减。

DE-ESS 的 dB 数值限定最大处理深度，检测不到相应事件时不会持续施加该深度。当前检测包络名义启动/释放为 1/78 ms，增益控制名义启动/释放为 1.4/92 ms。检测电平、增益平滑和处理顺序共同决定实际结果。图例是合成测试信号的两份完整预设，不是所有歌手都会得到的衰减曲线。

SOFT 另行比较快速与较慢的检测包络，用于减轻短促、突出的存在感事件。它不是 DE-ESS 的第二个上限，也不是把整段高频恒定压低的音质开关。

频谱图的阅读边界

频谱为小窗口显示：动态频谱从 24 个聚合频率点插值绘制，分析范围为 80 Hz–16 kHz；自适应窗口依据 90 Hz–10.5 kHz 的检测中心显示。高频合并为区域能量，插值用于连接图形，并不会增加测量本身的频率分辨率。显示范围限制不等于音频低通，窗口外的高频不会因此被额外删除。

RAW、CAL、OUT 用于比较三个信号位置的电平；曲线用于观察趋势和处理活动。它们不是校准声压计，也不能直接代替完整频响测量或绝对 LUFS 分析。

11. 延迟的样本数与时间

下表是当前处理器实际准备后读回的上报延迟。低延迟列关闭离线高音质预留；标准与高音质相同。低延迟模式若启用离线高音质偏好，也预留标准／高音质列的延迟。数值包含内部声音处理与干湿路径对齐。

采样率	低延迟：样本 / 时间	标准／高音质：样本 / 时间
44.1 kHz	12 / 0.272 ms	242 / 5.488 ms
48 kHz	13 / 0.271 ms	243 / 5.063 ms
88.2 kHz	25 / 0.283 ms	255 / 2.891 ms
96 kHz	26 / 0.271 ms	256 / 2.667 ms
176.4 kHz	50 / 0.283 ms	280 / 1.587 ms
192 kHz	52 / 0.271 ms	282 / 1.469 ms

换算关系为：毫秒 = 样本数 ÷ 采样率 × 1000。例：48 kHz 下 13 个样本约为 0.271 ms，243 个样本约为 5.063 ms。不同采样率的离散延迟来自实际准备与峰值定位，不应只对 48 kHz 数字作比例推算。

这些数值不包含音频接口、DAW 缓冲和其他插件。宿主 PDC 可以对齐已报告的路径，但不能取消实时监听已经发生的等待。更低的算法延迟也不代表没有滤波相位变化；当前版本不以改变已选音色的方式宣称整条链零延迟。

如果实际宿主显示与表格不同，先检查旧版是否仍被宿主加载、质量设置、离线预留和宿主延迟同步状态。保存工程后重新准备或重新打开插件，再读当前详情。

12. 常用处理流程

保留自然感的人声处理

1. 完成必要剪辑与明显片段音量修正后插入 LiVocal。先关闭暂时不需要的模块，建立处理前后的听感基准。
2. 根据录音低频决定低切，分别试听染色与压缩，再微调校准。
3. 选基础音色并少量倾斜；在伴奏中判断是否需要空气感、厚度或靠前。
4. 从适量的音色稳定与去齿音开始，比较辅音、元音和强弱句。必要时减少处理范围或关闭柔化。
5. 对代表性片段使用 AUTO，之后用 OUTPUT 微调并保存。

改变模块顺序



图 15. 简化信号关系；五模块内部仍有各自动态、滤波和补偿。内部输入平衡仅在动态模块启用时参与。

将智能处理移到润色之前，可先处理突出成分再调整润色；放到之后，则会对润色后的信号作判断。结果取决于音源和设置。重新排序后应重新检查声音、输出电平和 AUTO 是否需要重测。

并行混合与公平比较

用 MIX 减少完整处理声的比例，可保留更多原始表现。先在同一片段内设定混合，再调整 OUTPUT 或运行 AUTO；切换 A/B 时比较相近电平，避免只因更响就选择一个版本。

说明

这些流程是操作建议，不是固定预设或音质保证。它们不替代对完整人声、伴奏和最终导出的检查。

13. 故障排查

现象	检查与处理
DAW 找不到插件	确认安装对应格式、宿主支持该格式和系统版本；重新打开宿主并执行其插件扫描。Pro Tools 使用 AAX，AU 不会列为 AAX。
插件无声并提示激活	检查授权信息与管理器；确认付费许可或试用短期许可有效。试用的新宿主进程与刷新需要联网。
未提示激活，但表头无输入	检查轨道路由、静音、播放状态和插件前级信号；旁路或其他插件也可能已使信号静音。
旋钮变灰或似乎不作用	检查全局开关与所属模块。校准和稳幅为独立控制；SOFT 不在宿主自动化列表中。
AUTO 显示 NO SIG	播放有持续有效人声的片段，避免全局旁路和静音，再测量。仅有静音或过低信号不能完成有效比较。
AUTO 显示 LIMIT	输出已触及范围边界；检查输入增益、校准与链内处理。先使电平关系合理，再决定是否重测。
录音或切换后延迟不符合预期	查看当前质量、离线高音质预留、插件详情和宿主 PDC。停止播放后让宿主更新；必要时重新加载。
恢复工程后的范围与新实例不同	工程保留既有声音控制模式。先保留工程副本，阅读升级说明；不要仅因界面不同就覆盖原设置。
输出削波、处理过重或音色发闷	降低 OUTPUT；逐个比较模块，减少校准、音色稳定或去齿音范围，检查是否叠加了其他处理。
看不到工厂预设	确认安装当前完整安装包和对应格式的预设。Studio One 可刷新插件浏览器；Pro Tools、Logic 与 VST3 宿主使用不同预设格式。
打开的手册仍是旧版	关闭旧 PDF，从当前插件菜单重新打开，核对封面的产品版本。不要把阅读器已打开的旧标签页当作新文档。

建议

把问题缩小到一条轨、一个 LiVocal 实例和可重复片段，可帮助区分信号、授权、宿主与参数设置的问题。保留原工程再做简化测试。

14. 支持与联系

官网与下载：<https://www.jmechoaudio.com/>。问题反馈：

<https://www.jmechoaudio.com/#support>。支持邮箱：lijiabin124@outlook.com。

- 说明产品版本、macOS 或其他系统版本、DAW 名称与版本，以及使用的 AU、VST3 或 AAX 格式。
- 写出能够重复问题的操作顺序、当时设置，以及期望结果和实际结果。
- 截图应包含有关控件或错误信息；涉及授权时仅提供核对所需的订单信息。
- 不要发送密码、支付验证码、设备私钥或无关个人资料；提交音频前确认你有权分享。

建议

一段简短、可重复的故障描述比仅说“不能用”更有助于定位。音频问题可另附处理前后片段，并注明其他插件和自动化是否启用。

15. 许可与法律声明

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JMECHO Audio

LiVocal

Vocal Processor | User Guide

Listen first. Shape what needs to change.



Product version: 1.0.5

Manual revision: 2026.10-R3

Date: 2026-10-13

English edition

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Page numbers match the PDF reader. Select a chapter to navigate.

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1. Overview

LiVocal brings preamp character, dynamics, tone shaping and adaptive processing into one clear panel. Choose a base character, then refine brightness, body and dynamics. Enable individual modules or change their order as needed.

Gold, Silver, Pearl and Brass: four distinctive musical-style tonal curves, each offering a different starting point.

This guide uses a new 1.0.5 instance as its baseline: GOLD character, STANDARD quality, the physical interface and 100% interface size. Sound modules start enabled; TONE, AIR and BODY start at 0%, and MIX at 100%.

Where to use it

- Use it on a mono or stereo vocal track. Input and output channel counts match. The product has no external sidechain input.
- Perform necessary editing, noise treatment and large clip-level corrections before shaping and controlling the vocal. Choose the wider processing order for the source and task.

Note

LiVocal does not provide pitch correction, timing editing or automatic vocal comping. Setting AIR, BODY and TONE to 0% does not bypass the chain: other enabled modules and the selected base character still process audio.

Finding information

Task	Read
First session	Setup and licensing; Quick start
Find a control	Interface map; Parameter reference
Compare or save sounds	Toolbar and presets
Choose tracking or export settings	Quality, latency and recall
Troubleshoot	Troubleshooting

2. Setup and licensing

Item	Installer and host
Product version	LiVocal 1.0.5
System requirements	Use the requirements on the official download page for your platform. Your host may require a newer system.
Plug-in formats	AU / VST3 / AAX Native
Channels	Mono and stereo; no external sidechain.

Install the complete official package, then use the format your host supports: AU in Logic, AAX in Pro Tools, and VST3 in Studio One or REAPER as appropriate for the platform. The plug-in and shared licence manager have separate version numbers.

1. Download the matching installer from JMECHO, run it and complete the system prompts.
2. Open the JMECHO licence manager if it did not open automatically after installation.
3. Activate a paid licence or an available trial. Check Licence information in the plug-in.
4. Reopen the DAW, rescan if needed and insert LiVocal on an audio track.
5. Play the vocal, check the input/output meters, and save.

Paid licensing and offline use

LiVocal uses a perpetual paid licence, not a subscription. Each licence provides 2 device slots and up to 3 self-service migrations per rolling 365 days. First activation and migration need the separate helper online. A valid perpetual licence stored locally allows offline use on the covered device.

Reinstalling the system or deleting device keys/licence files can require verification again. Contact support for lost devices, recovery or migration limits; do not delete licensing data to attempt a reset.

Seven-day trial

An available trial lasts 7 consecutive 24-hour periods from first server issuance, with no automatic charge. Trial start, each new host-process launch and lease renewal need Internet access. A lease lasts at most 24 hours, never beyond trial expiry.

Caution

Without a valid paid licence or trial lease, LiVocal cannot pass audio. Check licensing before critical or offline work and export. The helper needs network permission.

3. Quick start

Loop a vocal passage containing quiet delivery, normal phrases and stronger syllables. Use the following sequence as a starting point, then refine it for the recording.

1. Start with a new instance, GOLD, STANDARD quality and MIX at 100%. Check for signal on the RAW and OUT meters during playback.
2. Audition PRE first. Compare COLOR and COMP separately. Compare LOW CUT when retaining low fundamentals matters or the recording was already filtered.
3. Adjust CALIBRATE as input calibration, not a compression-amount control. At 50% its calibration gain is zero; moving either side changes the level driving the chain.
4. Compare the four base characters, then make a small TONE adjustment. At 0% the selected character remains; negative values darken it and positive values brighten it.
5. Set AIR, BODY and MID as needed. Then use EQ RANGE, DE-ESS and SOFT to control uneven or prominent components.
6. Reduce MIX for a parallel blend. Hover over OUTPUT, select AUTO, and keep playing active vocal until the measurement finishes.
7. After AUTO finishes, compare the processed sound and refine OUTPUT by ear. Save a preset in the host plug-in header for reuse, and save the session as well.

Note

AUTO compares input and output over about two seconds of active audio and writes OUTPUT once. It is not continuous gain riding or an absolute LUFS target.

Tip

Listen to each change before keeping it. Enabled defaults are a starting point; switch modules off individually to identify their contribution.

4. Interface map



Figure 1. LiVocal 1.0.5 physical interface at new-instance settings. Bottom numbers locate the corresponding modules; detailed figures identify controls. Readouts illustrate the captured state.

Area	Entries	Purpose
PRE	1-4	Low cut, coloration and preamp compression
DYNAMICS	5-7	Dynamics module, global calibration and stabilisation
CHARACTER	8-11	Four base characters, tone tilt and WARM
POLISH	12-15	AIR, BODY and MID
ADAPTIVE	16-19	EQ range, de-essing and softening
MASTER	20-23	MIX, OUTPUT, AUTO and global power

Control operation and feedback

Drag vertically: up increases and down decreases. Hold Command or Shift for finer changes, without jumps when you press or release the modifier. Click a value to type; Return confirms and Esc cancels. Manual steps are 1% or 0.1 dB; automation keeps its original precision.

The module switch enables its stage. Drag a three-line handle to reorder the five sound modules; MASTER stays last. RAW is input, CAL is calibrated and internally balanced level, and OUT is output. The displays show activity and cannot edit EQ bands.

Note

The simple and physical views use the same sound parameters. MID enables midrange presence; EQ RANGE sets the permitted adaptive EQ correction. Switching views does not change the current sound.

5. Parameter reference

Numbers match the interface figures. Defaults refer to a new 1.0.5 instance; session recall, presets and A/B may restore other values. Module switches bypass their sound processing. CALIBRATE and STAB are independent controls; see their entries.

PRE



Figure 2. PRE controls 1-4. LOW CUT precedes preamp coloration and compression; switches are enabled in this view.

1. PRE ON

Enables or bypasses the PRE module.

Range: On / Off / Default: On / Unit: None

Effect: When enabled, LOW CUT, COLOR and COMP follow their own settings.

Conditions: Global power must be on. Disabling PRE leaves the other modules available.

2. LOW CUT / CUT

Applies a fixed high-pass filter before preamp processing.

Range: On / Off; 90 Hz, 12 dB/oct / Default: On / Unit: Switch

Effect: Reduces low-frequency rumble and energy, including useful low-frequency vocal content.

Conditions: Active when PRE is enabled.

Tip: If the recording is already filtered, compare the resulting body.

3. COLOR

Enables the preamp coloration stage.

Range: On / Off / Default: On / Unit: None

Effect: Changes harmonic character without also enabling COMP.

Conditions: Requires PRE. COLOR and COMP are independent.

4. COMP

Enables preamp compression and its associated dynamic resonance control.

Range: On / Off / Default: On / Unit: None

Effect: Controls preamp dynamics and selected midrange resonances without also enabling COLOR.

Conditions: Requires PRE; includes fixed 900 Hz / 3 kHz dynamic treatment.

Tip: Compare COMP and COLOR separately to identify their contributions.

DYNAMICS



Figure 3. DYNAMICS area, 5-7. Entry 6 calibrates the chain input; entry 7 is independent post-module stabilisation.

5. DYNAMICS ON

Enables or bypasses the dynamics control module.

Range: On / Off / Default: On / Unit: None

Effect: Enables multiband control, associated vocal adaptation and internal input balancing.

Conditions: CALIBRATE and STAB remain independent when this module is off.

6. CALIBRATE

Sets the level driving the sound chain and applies inverse calibration gain afterwards.

Range: 0-100% / Default: 50% / Unit: %

Effect: At 0%, 50% and 100%, input calibration is -12, 0 and +12 dB. Higher drive can change compression and nonlinear processing; inverse gain does not undo those changes.

Conditions: This is global input calibration and remains active with DYNAMICS off.

Tip: Start at 50%, watch CAL and listen.

Caution: This is not compression amount or an absolute loudness target.

7. STABILIZER / STAB

Tracks and compensates dynamic level loss from preamp and dynamics processing.

Range: On / Off / Default: On / Unit: None

Effect: Compensation is applied once after the movable modules, separating compression loss from character-shaping changes.

Conditions: Independent of the module switch. It is not OUTPUT AUTO or a continuous LUFS lock.

CHARACTER

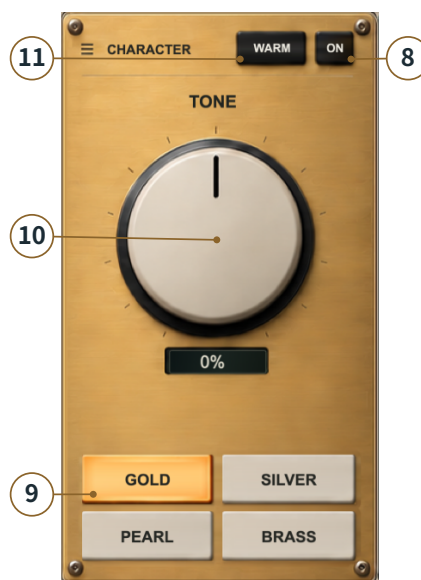


Figure 4. CHARACTER controls 8-11. TONE at 0% preserves the selected base character.

8. CHARACTER ON

Enables or bypasses base-character and tone-tilt processing.

Range: On / Off / Default: On / Unit: None

Effect: When off, the character selection, WARM and TONE do not process audio in this module.

Conditions: Other modules remain independent.

9. Base character

Selects a fixed base-character response.

Range: GOLD, SILVER, PEARL, BRASS / Default: GOLD / Unit: Choice

Effect: Compare the four responses on the same source; no choice is universally correct for every vocalist.

Conditions: Requires CHARACTER. WARM also affects the base response.

Tip: Store two candidates in A/B and compare at similar output levels.

10. TONE

Tilts tonal balance after the selected base character.

Range: -100% to +100% / Default: 0% / Unit: %

Effect: Negative values darken the balance; positive values brighten it. At 0% this tilt is neutral. Its measured endpoints are asymmetric rather than mirrored standard shelves.

Conditions: New instances use the measured tilt, independently of the four base characters. Requires CHARACTER.

Tip: Check sibilance, body and mix placement after a small change.

Caution: The tilt itself adds no harmonics or noise; other parts of the chain can still be nonlinear.

11. WARM

Switches the definition used for the selected base character.

Range: On / Off / Default: On / Unit: None

Effect: On selects the warm definition; off selects the original definition. This changes sound, not just panel colour.

Conditions: Requires CHARACTER. Recheck level and tone after switching.

POLISH

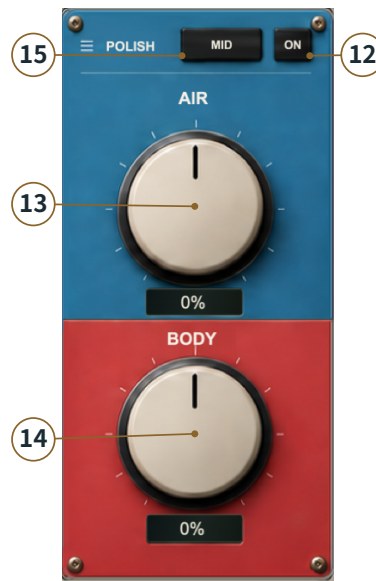


Figure 5. POLISH controls 12-15. AIR and BODY both have positive and negative ranges.

12. POLISH ON

Enables or bypasses POLISH.

Range: On / Off / Default: On / Unit: None

Effect: Controls whether AIR, BODY and MID process the signal.

Conditions: Their settings are retained while the module is off.

13. AIR

Adds or reduces high-frequency air.

Range: -100% to +100% / Default: 0% / Unit: %

Effect: Positive values combine high-frequency EQ and harmonic polish; negative values attenuate high frequencies. At 0% this control is neutral.

Conditions: Requires POLISH. DE-ESS remains independent.

Tip: Check AIR and DE-ESS together on a representative passage.

14. BODY

Adds or reduces body and low-frequency weight.

Range: -100% to +100% / Default: 0% / Unit: %

Effect: Positive values add body with associated polish; negative values reduce it.

Conditions: Requires POLISH. LOW CUT also affects the resulting low-frequency balance.

Tip: Check congestion in the mix as well as in solo.

15. Midrange presence / MID

Enables a fixed midrange presence EQ.

Range: On / Off; 2000 Hz / Default: On / Unit: Switch

Effect: Adds midrange presence without extra harmonic generation.

Conditions: Requires POLISH.

Tip: Compare this switch if vowels become too prominent.

ADAPTIVE



Figure 6. ADAPTIVE controls 16-19. The dB values set processing ranges or limits, not a constant reduction.

16. ADAPTIVE ON

Enables or bypasses ADAPTIVE.

Range: On / Off / Default: On / Unit: None

Effect: Controls EQ RANGE, DE-ESS and SOFT processing.

Conditions: Other dynamics and character modules remain independent.

17. EQ RANGE

Sets the permitted range of adaptive tonal correction.

Range: 0-8 dB / Default: 3.0 dB / Unit: dB

Effect: Higher values allow stronger correction. At 0 dB this tonal correction is disabled; DE-ESS and SOFT remain separate.

Conditions: Requires ADAPTIVE. New instances use the zero-capable range.

Tip: Increase gradually and judge vowel naturalness and phrase-to-phrase consistency.

18. DE-ESS

Sets the maximum reduction for dynamic high-frequency smoothing.

Range: 0-8 dB / Default: 3.0 dB / Unit: dB

Effect: Allows attenuation of prominent high-frequency content. At 0 dB this process is off; the setting is not a constant cut.

Conditions: Requires ADAPTIVE; labelled DE-ESS in the simple view.

Tip: Check sibilant phrases, then compare overall brightness.

19. SOFT

Enables dynamic softening in the presence region.

Range: On / Off / Default: On / Unit: None

Effect: Softens prominent presence-region transients, with a different scope from DE-ESS.

Conditions: Requires ADAPTIVE. SOFT is not a host-automatable parameter.

MASTER



Figure 7. MASTER controls 20-23. Entry 22 points to the OUTPUT hover area for AUTO; the button appears on hover or during measurement. Global power, 23, is in the top bar.

20. MIX

Sets the blend between dry and processed audio.

Range: 0-100% / Default: 100% / Unit: %

Effect: At 0% the blend uses dry audio; 100% uses the processed path. Intermediate values create a parallel blend. Internal timing and level compensation apply.

Conditions: OUTPUT still sets final level. MIX at 0% is not global bypass.

21. OUTPUT

Adjusts final plug-in output gain.

Range: -18 to +18 dB / Default: 0.0 dB / Unit: dB

Effect: Raises or lowers final level. AUTO changes this value once.

Conditions: Applied after the blend; it does not lock absolute LUFS.

Tip: Account for level differences when comparing bypass, and check downstream headroom.

22. AUTO output matching

Briefly compares input and output, then adjusts OUTPUT once.

Range: One-shot measurement; result limited by OUTPUT range / Default: Idle / Unit: Action

Effect: After approximately 2 seconds of active audio, adjusts output gain once from the input/output comparison. Without enough active signal, it stops after approximately 15 seconds.

Conditions: Requires valid licensing, global power and playback. Keep settings and automation stable during measurement.

Tip: DONE means applied; LIMIT means the output boundary was reached; NO SIG means no valid measurement.

Caution: It measures the current passage only. Reassess after changing character, order, MIX or source.

23. Global power

Switches the overall plug-in between processing and bypass when licensed.

Range: On / Off / Default: On / Unit: None

Effect: When off, the dry path matches the reported latency, with a smoothed transition.

Conditions: Bypass keeps processing state warm. It does not unload the plug-in or reduce CPU, and does not bypass licensing.

6. Compare, presets and options

Control	Operation and result
A / B	Select either sound slot, each with its own sound settings and module order.
COPY	Copy the current sound into the other slot.
← / →	Undo or redo sound edits. Disabled arrows retain the same printed colour.
EN / 中	Shows the language that clicking will switch to.
≡	Groups Quality & Export, Interface, and Help & Info.

Recall presets from the host

The plug-in options menu no longer manages presets. Use the host plug-in header to recall factory sounds and save your settings, and save the session too. The same recipes are supplied in the preset format appropriate to AU, VST3 or AAX.

Name	Starting point
01 Gold Default	A balanced Gold starting point.
02 Pearl Gentle	Gentle shaping for natural phrasing.
03 Silver Air	More top-end openness for a darker recording.
04 Brass Body	More body for a thinner recording.
05 Gold Forward	More midrange presence in a busy arrangement.
06 Pearl Open	An open Pearl tone with gentle correction.
07 Warm Close	A warmer, intimate starting point.
08 Bright Control	Brightness with control over prominent highs.
09 Sibalance Soft	A de-essing starting point; adjust for the recording.
10 Brass Parallel	A 70% wet Brass blend.
11 Clean Shape	EQ-focused shaping with PRE colour and compression, DYNAMICS and ADAPTIVE off.
12 Dry Detail	An 80% wet blend for detail and original expression.

Audition first

A preset is a starting point, not a diagnosis of the recording. Audition the same phrase, check calibration, sibilance and output, then adjust for the source. All factory presets use 0 dB output gain.

More options

Group	Contents
Quality & Export	Select Low Latency, Standard or High Quality, and enable high-quality offline export.
Interface	Switch directly between Simple and Physical, and choose interface size.
Help & Info	Getting started, User manual, Plugin details and Licence information.

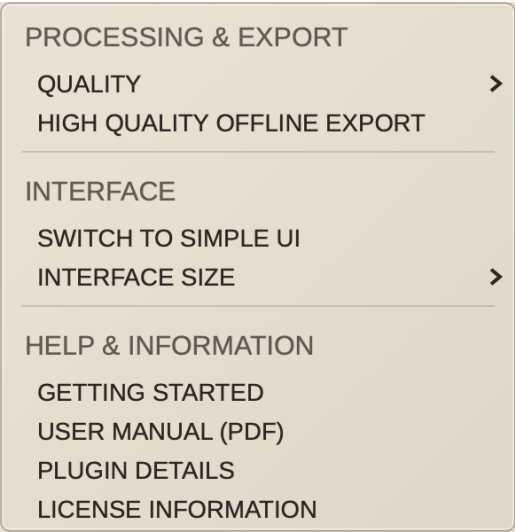


Figure 8. Grouped More Options. Access factory presets through the DAW.



Figure 9. The Simple view uses the same sound controls as the Physical view.

Sound presets and A/B comparisons retain the current view, language, quality and offline-export preference. A saved session can restore those preferences too; each host has its own rules for saving plug-in presets.

7. Quality, latency and recall

Quality option	Use
LOW LATENCY	Use when processing cost and lower algorithmic latency matter. Reported latency can still reflect an offline high-quality reservation.
STANDARD	Default for a new instance and a starting point for normal processing and comparison.
HIGH QUALITY	Uses higher processing quality and cost. Check available system headroom.

LiVocal reports latency to the host and aligns blend and bypass paths. A quality name does not specify total recording latency. The value depends on sample rate, mode, offline high-quality reservation and host handling. Check Plugin details and the host.

Recording and monitoring

For lower monitoring latency, select Low Latency and check that high-quality offline export is off. Quality modes use different processing; they are not identical-sounding speed settings. Total recording latency also includes the host buffer, audio interface and other plug-ins.

High-quality offline export

This preference is off by default. When enabled, the plug-in uses high quality if the host calls it in non-realtime mode. Realtime playback or realtime export uses the manually selected quality. Enabling the preference reserves high-quality latency and notifies the host.

Caution

After changing latency-related settings, stop playback and confirm the host updates compensation. If the menu reports that latency sync is pending, allow the host to prepare or reload the plug-in before offline export. An export label alone does not guarantee a non-realtime call.

Recalling an existing session

An existing session may retain its original tone-control algorithm and stability range. The interface binds the appropriate controls for the saved state, so recalled ranges can differ from a new instance. Explicit upgrade actions in the menu change those sound interpretations.

Caution

Save a session copy and note automation before upgrading. Existing automation is not automatically converted to a new sound meaning; the old and new stability ranges use different parameters. Compare the result before adopting it.

8. Frequency, level and response

Hz and kHz describe frequency. dB describes a level ratio; dBFS is a signal level relative to digital full scale. A percentage is a control position, not a compression ratio, frequency or sustained gain reduction.

TONE: brightness and phase

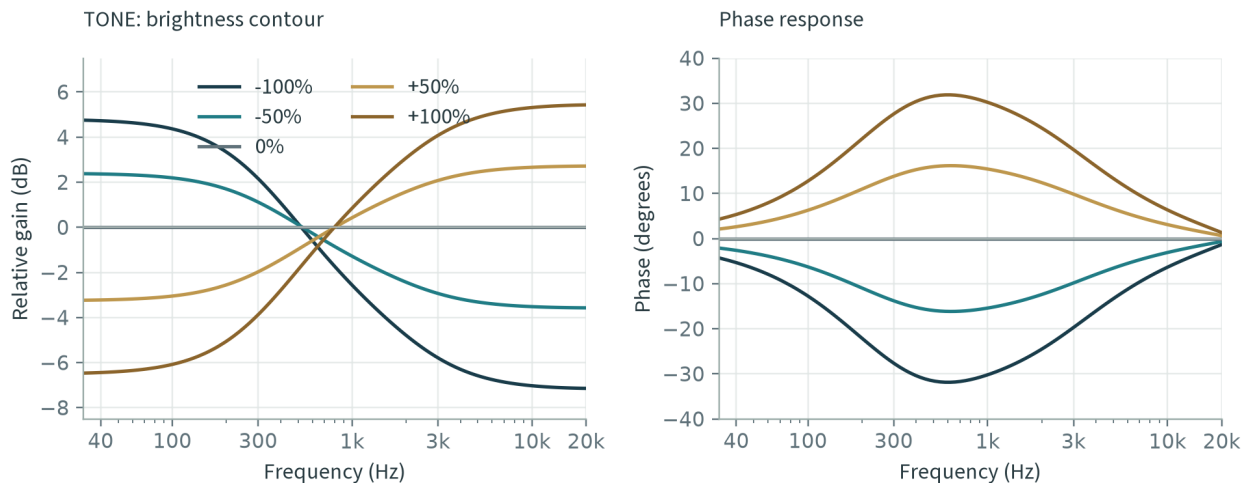


Figure 10. Measured TONE magnitude and phase at 48 kHz, using settled sine waves with 0.1 peak amplitude. This isolates TONE; the four base characters, PRE and other modules are excluded.

0% is neutral within this stage. Negative values favour the low end and reduce the top end; positive values tilt toward brightness. The approved measured endpoints are not exact mirror images. Intermediate settings blend toward the corresponding endpoint.

Phase response is the angular shift of each output frequency relative to its input. It is distinct from the delay reported to the host and does not describe the whole recording path. TONE adds no harmonics or noise of its own; other enabled stages can still alter harmonics and dynamics.

MID: broad midrange presence

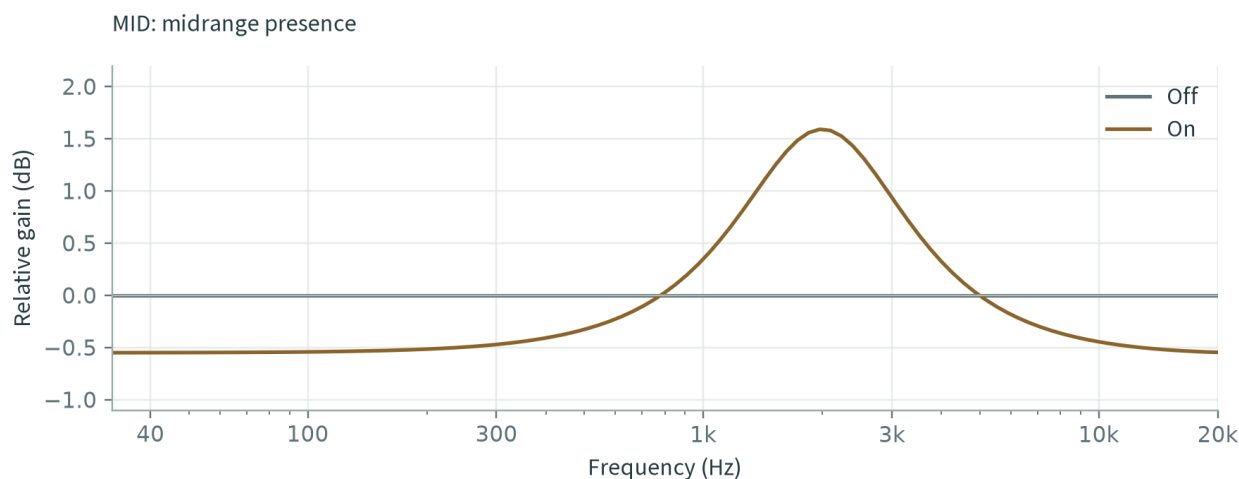


Figure 11. Measured settled MID response at 48 kHz. Off is neutral; On combines broad midrange lift with fixed level compensation.

The centre is 2 kHz with Q 0.78. The bell contributes +2.142 dB and is followed by -0.55034 dB level compensation, giving about +1.59 dB net lift at the centre. Q describes width: this is broad shaping, not a narrow resonance-removal control.

MID operates inside POLISH with a smoothed switch transition and no added look-ahead. Compare intelligibility and vowel prominence in the mix.

CUT: frequency and slope

PRE CUT is a fixed 90 Hz high-pass with a 12 dB/oct slope and Q about 0.707. It does not abruptly remove everything below 90 Hz. Attenuation increases toward lower frequencies; 12 dB/oct is the asymptotic slope, not a fixed 12 dB cut at every bass frequency.

9. Understanding calibration and dynamics

CALIBRATE: change the level driving the chain

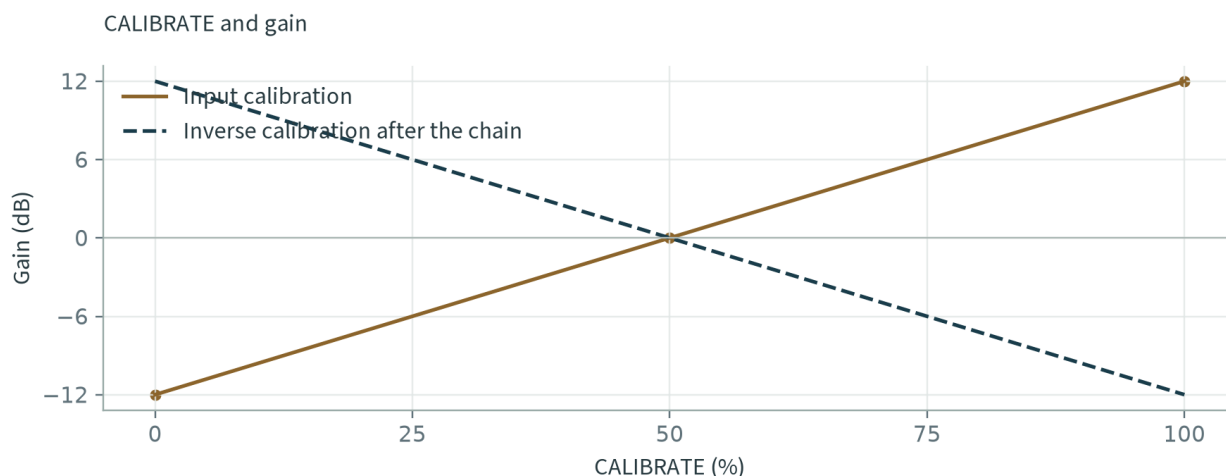


Figure 12. Exact calibration gain mapping. The solid line is input calibration; the dashed line is its inverse after the chain. These are not whole-chain loudness curves.

CALIBRATE	Input calibration	Inverse calibration
0%	-12 dB	+12 dB
25%	-6 dB	+6 dB
50%	0 dB	0 dB
75%	+6 dB	-6 dB
100%	+12 dB	-12 dB

Input calibration is $0.24 \times$ the control percentage – 12 dB, with about 25 ms gain smoothing. Its inverse compensates that linear gain step only. Compression, colour and adaptive stages have already responded to the changed input, so it does not restore the original sound. STAB and OUTPUT AUTO remain separate.

DYNAMICS: four detection regions

Four dynamics detection regions

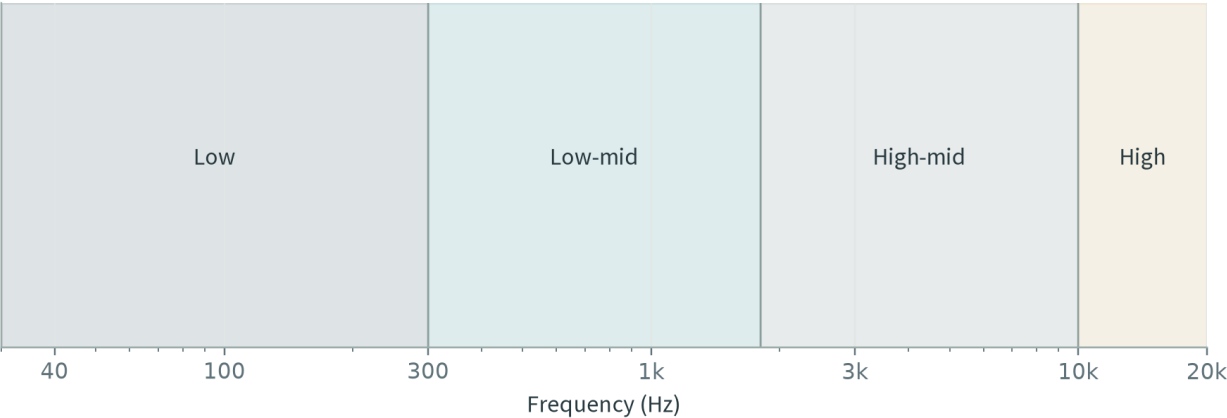


Figure 13. Illustration of detection regions, divided at 300 Hz, 1.8 kHz and 10 kHz. Colours identify regions; they are not gain curves or brick-wall audio splits.

Detector region	Nominal attack	Base release
Low	18 ms	160 ms
Low-mid	10 ms	110 ms
High-mid	5 ms	80 ms
High	2.5 ms	55 ms

These are the current internal nominal envelope settings, not four independent user controls. A 4:1 base ratio is combined with broad knees, detector calibration, strength mapping and fast/slow envelopes. Sustained and brief events can release differently; the table is not a measurement of how quickly the complete output reaches a particular percentage.

Attack describes detector or gain-control speed; it is not a delay added to wait for future samples. Judge phrase movement, consonants and breaths alongside the meter.

10. Adaptive processing, de-essing and displays

EQ RANGE: permitted correction, not a fixed EQ

Adaptive EQ uses 12 detector centres: 90, 140, 220, 350, 550 and 850 Hz, then 1.3, 2, 3, 4.5, 6.8 and 10.5 kHz. It compares neighbouring spectral energy with its tonal target and smooths the corrections. These centres are not 12 editable static EQ bands.

New instances give both EQ RANGE and DE-ESS a 0–8 dB range, so 3.0 dB has the same knob angle. EQ RANGE at 0 returns EQ targets smoothly to neutral; it does not turn off DE-ESS or SOFT. A 3 dB setting does not impose a constant 3 dB change at every frequency across the whole chain.

DE-ESS: the limit and the actual reduction

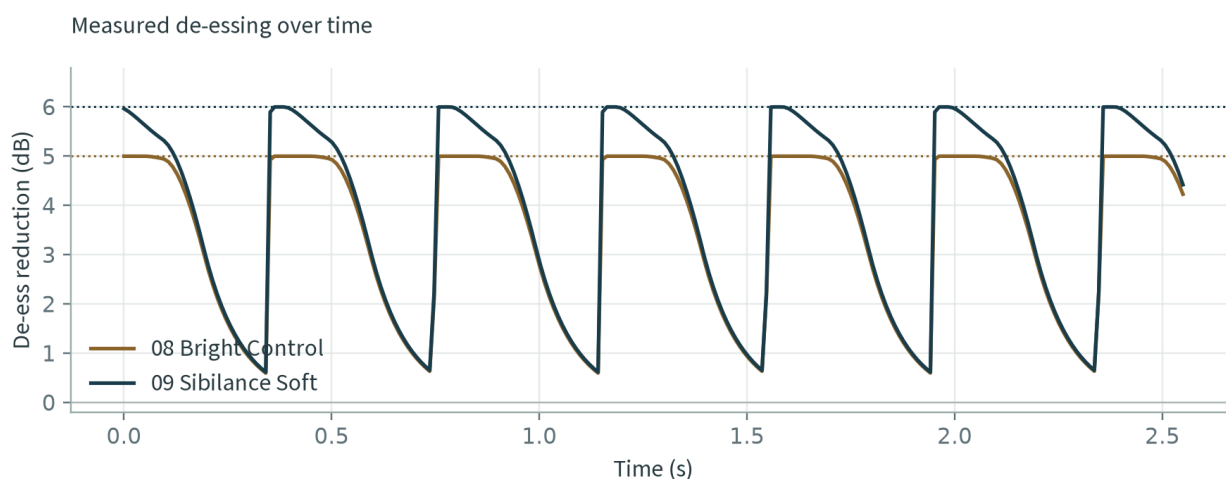


Figure 14. Actual de-ess readings for presets 08 and 09, capped at 5 and 6 dB. Measured at 48 kHz in 512-sample blocks, after settling, with intermittent 8.2 kHz bursts over a 300 Hz base. Dotted lines are configured limits; solid lines are reduction sampled at each block end.

DE-ESS limits maximum depth; that depth is not continuously applied when no matching event is detected. Nominal detector attack/release is 1/78 ms and gain-control attack/release is 1.4/92 ms. Level, smoothing and processing order determine the result. The figure shows two complete presets on a synthetic fixture, not a universal response for every singer.

SOFT separately compares faster and slower detector envelopes to soften brief, protruding presence events. It is neither a second DE-ESS limit nor a constant whole-treble reduction switch.

Reading the spectrum displays

The mini dynamics spectrum interpolates 24 aggregated frequency points covering 80 Hz–16 kHz. The adaptive window displays detector centres from 90 Hz to 10.5 kHz. High-frequency energy is grouped into regions; connecting the plot does not increase analysis resolution. Display limits are not audio low-pass filters and do not remove frequencies outside the window.

RAW, CAL and OUT compare levels at three signal positions; curves show trends and processing activity. They are not calibrated sound-pressure meters, full transfer-function measurements or absolute LUFS analysis.

11. Latency in samples and time

The table reads the actual delay reported by the current processor after preparation. The low-latency column disables the offline-HQ reserve; Standard and High Quality report the same delay. Low Latency with offline HQ enabled also reserves the Standard/HQ delay. Values include internal processing and dry/wet alignment.

Sample rate	Low latency: samples / time	Standard/HQ: samples / time
44.1 kHz	12 / 0.272 ms	242 / 5.488 ms
48 kHz	13 / 0.271 ms	243 / 5.063 ms
88.2 kHz	25 / 0.283 ms	255 / 2.891 ms
96 kHz	26 / 0.271 ms	256 / 2.667 ms
176.4 kHz	50 / 0.283 ms	280 / 1.587 ms
192 kHz	52 / 0.271 ms	282 / 1.469 ms

Milliseconds = samples ÷ sample rate × 1000. At 48 kHz, 13 samples are about 0.271 ms and 243 samples about 5.063 ms. Sample delays at other rates come from actual preparation and discrete peak location; do not infer them only by scaling the 48 kHz value.

These figures exclude the interface, DAW buffer and other plug-ins. Host delay compensation aligns reported paths but cannot undo waiting that already occurs during live monitoring. Low algorithmic latency also does not imply zero filter phase shift; this version does not claim a zero-delay complete chain by changing the approved sound.

If the host shows a different value, check whether it still has an old loaded version, the quality mode, offline reserve and delay-sync status. Save the session, then reprepare or reopen the plug-in and check its current details.

12. Practical workflows

A natural vocal starting point

1. Insert LiVocal after necessary editing and major clip-gain corrections. Disable unneeded modules and establish a before/after reference.
2. Choose LOW CUT for the source, audition COLOR and COMP separately, then refine CALIBRATE.
3. Choose the base character and a modest TONE tilt. Evaluate AIR, BODY and MID with the backing track.
4. Use moderate EQ RANGE and DE-ESS, comparing consonants, vowels and different delivery levels. Reduce ranges or turn SOFT off if needed.
5. Run AUTO on a representative passage, refine OUTPUT, and save.

Changing module order

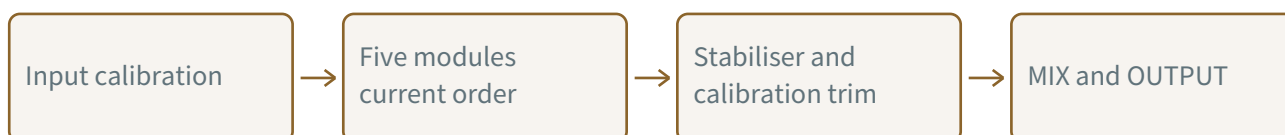


Figure 15. Simplified signal relationship. Modules contain their own dynamics, filtering and compensation. Internal input balancing is used with DYNAMICS enabled.

Placing ADAPTIVE before POLISH treats the signal before polish; placing it afterwards evaluates the polished signal. Results depend on the source and settings. Recheck sound, level and whether AUTO needs another measurement after reordering.

Parallel blend and useful comparison

Lower MIX to retain more of the original performance. Set the blend on the same passage before adjusting OUTPUT or running AUTO. Compare A/B at similar levels so loudness does not dominate the decision.

Note

These workflows are suggestions, not fixed presets or quality guarantees. Check the full vocal, backing track and final export.

13. Troubleshooting

Symptom	Check and action
Plug-in not listed	Check the installed format and host/system requirements. Reopen the host and rescan. Pro Tools uses AAX; AU will not appear as an AAX insert.
Silent with activation prompt	Check Licence information and the manager. Verify the paid licence or trial lease. A fresh trial host process and renewal require Internet access.
No input meter activity	Check routing, mute, transport and signal before the plug-in. Other inserts can also silence the path.
Greyed or inactive control	Check global power and the owning module. CALIBRATE and STAB are independent. SOFT is absent from the host automation list.
AUTO shows NO SIG	Play active vocal, avoid global bypass and mute, then retry. Silence or very low signal cannot complete a valid comparison.
AUTO shows LIMIT	OUTPUT reached its boundary. Check input level, CALIBRATE and processing. Correct the level relationship before retrying.
Unexpected latency	Check quality, offline reservation, Plugin details and host PDC. Stop playback and let the host update; reload if necessary.
Recalled ranges differ	The session retains its sound-control mode. Save a copy and read the upgrade notes before changing the original settings.
Clipping or overprocessing	Lower OUTPUT, compare modules individually, reduce drive or correction ranges, and check processing in other inserts.
Factory presets are missing	Check the current complete installer and preset format. Refresh the plug-in browser in Studio One if needed; Pro Tools, Logic and VST3 hosts use different preset formats.
The manual is out of date	Close the old PDF and reopen User manual from the current plug-in. Check the cover version; an existing reader tab may still show an older document.

Tip

A single track, one LiVocal instance and a repeatable passage help separate routing, licensing, host and parameter issues. Keep the original session before simplifying it.

14. Support and contact

Website and downloads: <https://www.jmechoaudio.com/> . Feedback: <https://www.jmechoaudio.com/#support> . Support email: lijiabin124@outlook.com .

- Include the product version, operating system, DAW name and version, and AU, VST3 or AAX format.
- Describe reproducible steps, relevant settings, the expected result and the observed result.
- Include the relevant controls or error message in screenshots. For licensing issues, provide only the order details needed for verification.
- Do not send passwords, payment codes, device private keys or unrelated personal information. Confirm that you may share any submitted audio.

Tip

A short reproducible description helps diagnosis. For an audio issue, a before/after excerpt can help; state whether other plug-ins and automation were active.

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