Ear Fatigue / Auto-Dynamics Mastering Plugin

MVP (Minimal Viable Product) Checklist

☑ **Idea Concept:**

- An audio visualisation app that helps producers maintain appropriate audio levels in their mixing as they go along to protect dynamics, and encourage them to adjust their playback device volume level accordingly instead entering 'perceived loudness' at the mixing stage.

☑ **Core Functionality - 3 Core Features:**

1. Dynamic range visualisation with traffic light colours
2. Algorithm for detecting dynamics vs perceived dynamics of audio signal
3. Warning indicator of moving from loud to 'perceptively loud', e.g. when too much reduction in dynamic range is occurring (i.e. over-compression and squashing of overall dynamics).

☑ **Technical Requirements:**

1. Loudness metering (perceived loudness vs actually loudness, i.e. algorithm must approximate the point at which audio has likely undergone compression/limiting, and output by how much the audio signal has been compressed/limited, represented by a number or number rang)
2. Visual representation of audio signal compression/limiting amount, based on a set standard range, such as mastering standards, or even input reference tracks. traffic light colour-based, with red being warning of over-compression, with green being within range.
3. Audio signal stream real-time processing, doesn't have to be accurate to sub-second timings
4. Simple user interface with Bypass button, mastering standard selection, reference track selection for dynamic range.
5. A visual representation of when the user should adjust the listening volume, by how much and in which direction, up or down. Perhaps a message that ear fatigue is likely, and to turn audio level down or rest.

☑ User Interface sketch

☐ Development phases

☑ **Pain-points:**

- Mixes always sound flat and overly compressed by the time I complete the production phase. How do I keep my ear's fresh, and how do I know if I'm squashing too much the dynamic elements of my music just because i've lost objectivity, due to ear fatigue.
- Mixing and mastering at the end of creating my music, I am almost uninterested in it by then, and I've lost interest in what I originally created, so I need to incorporate mixing and mastering into the creative process. However, it is difficult to know if i'm listening to actual loudness (large dynamic range), verse 'perceived loudness' (reduced dynamic range). There are sweet spots for most audio and music mediums, but how do I keep a sense of where the sweet spot is, while composing and producing music?
- How do I set my optimal listening volume, especially with headphones, based on my mood, tiredness, etc. Sometimes I want to listen loud and get into the music, but that's when I usually loose the most objectivity in dynamics and end up giving myself a huge amount of work at the end to do.
- The "Loudness wars" is pushing for 'perceived loudness' over actual dynamic range, which I think is where music is more enjoyable and lively. I would just like to know where the sweet spot for listening and enjoying music is, where I can be sure my music is actually dynamic and creates impact, rather than guessing how dynamic my music is.

☑ **Personal Observations:**

- Listen to Peter Gabriels - Sledgehammer (2012 Remaster). At the start, a panflute or panpipe instrument plays at a certain volume. I noticed that adjusting my listening levels at this point, made the rest of the track sounds absolutely amazing, full of life and impact. It's as if the producers put this sound at this level to calibrate the listener's ear to the sound level they wanted the music to be played at. I.e. It's meant to be loud and brash, like listening in the front row to performers play at a popular New Orleans music jazz cafe.

☑ **Product Goals: What does it achieve?**

- Simplify mixing/mastering for producers with limited experience
- Maintain dynamic range while staying within industry standards
- Reduce ear fatigue through better monitoring
- Keep producers in creative flow while ensuring technical quality

MVP - In more Detail:

☑ **Core Problem Statement:**

- Producers struggle to maintain objectivity about dynamic range while in creative flow, leading to either over-compressed mixes or extensive rework later. Different listening volumes affect frequency perception.

☑ **Unique Solution:**

- Visual feedback system using color indicators to maintain appropriate dynamic range without interrupting workflow

☑ **Key Features**:

Dynamic range visualization, frequency spectrum analysis, and industry standard presets

- Optimal listening levels are needed to maximise objectivity in dynamics perception, but it's difficult to know what these are.
- Types of users (musicians/producers) vary greatly - technical knowledge gap.
- Physical and Mental state may have an on dynamics perception
- Length of current listening session impacts ear fatigue and objectivity.
- Playback medium (headphones vs. speakers) impacts dynamics perception.

☑ **What are your personal observations:**

- Preferred listening volumes for mixing and mastering vary with may factors, including time of day, mental/physical fatigue, background noise, individual user preferred listening levels, etc.

☑ **Current way of doing things:**

- Use a Dynamic Loudness Metering plugin that shows how dynamic the track is, based on 'Perceived Loudness/LUFS'. The problem is that this is not actual loudness as a measure of dbSPL and therefore the user doesn't know if their ears are being 'tricked' into the actual dynamic range of the track, especially as listening to music at different volumes can distorted perceived loudness, adding to the problem of the loudness wars.
- Put a limiter and/or compressor on the master audio channel and hope that the track doesn't 'push' too hard into 'the red', reducing the overall dynamic range.

- Put a soft clipper on individual channels to minimise digital clipping, and hope it doesn't sound too distorted as the audio signal is push into the 'red'.
- Focuses on maximising psycho-acoustic loudness, instead of retaining dynamics and offering the user to instead turn their speakers up or down in order to retain objectivity. Again, the Fletcher-Munson curve shows that listening volume is important in keeping music dynamic yet impactful!
- Has set listening levels for mixing and mastering, but still doesn't account for individual preferences and listening fatigue levels.

☑ **Proposed changes to current way of doing things:**

- Incorporate more enjoyment into the production phase
- Consider individual listening AND playback devices for optimal listening level, via dynamic range feedback.
- Focus on maximising actual dynamic range, and remove the need the mastering stage in music and other media.

☑ Important Notes:

I really just want a way of knowing if I am 'perceiving' the music as loud 'Loudness perception/LUFS, verses the music is actually loud (equivalent dbSPL). The later ensuring ear fatigue is kept at a minimum an objectivity in the music is kept as dynamic (difference between loudest and quietest sound as possible. I want to be able to produce and mix my music, without worry that my auditory perception is being compromised, helping to produce music more quickly, by finding MY preferred listen levels - levels that will translate best across to other playback devices.

Motivation (5 Whys?) Checklist

☑ Does this relate to my 10-year goals?

Yes, I want to learn skills that I think will still be valuable in 10 yrs. I believe coding and creating software will be important in 10 years, and my idea of a ear-fatigue tool will help professionals maximise the time they can spend mixing and mastering music.

☑ Does this relate to my 1 year and 3 month goals?

Yes, I want to learn how to code and want an audio plugin that will help me get back into the flow of producing music, without worrying about mixing and mastering, production areas that I really don't want to spend time on, as I tend to lose perspective listening for long periods of time. I want to know the optimum listening level, and realise when my objectivity has been compromised, so I can call it a day.

☑ Does something like this already exist?

Yes, there are dynamics metering tools, but I want something to help me set my preferred listen levels easily, and I can't find anything that simple. They only focus on 'perceived loudness' and not 'actual loudness', which i think is more important when producing music. We can care about perceived loudness at the end, but I want to know if I'm perceiving things as loud because they are, or just because my ears are adjusting to constant levels in audio (ear fatigue)

☑ Does it seem worth the time, money or effort? (two times rule)

It seems like a great project for me to learn how to code, as well as learn deeper maths and science behind audio plugins. I would do this two times, because I really want to know if my hypothesis about ear fatigue and perceived-loudness is correct.

☑ Can you take on this project alone?

Yes, I can learn how to code, and I think I can learn the maths and science behind mastering levels, from my mastering and psychoacoustics books I have read.

☑ Would you still create it even if no-one cared, and you were the only one to use it?

Yes, I would get pleasure from the sense of building something on my own, as well as pleasure in figuring out why I find mixing and mastering so difficult in music production.

Sources of Inspiration

Break down 3 similar projects structure or characteristics

1. **Izotope Insight 2 £47**
    a. Loudness metering tool with multiple modules
    b. Displays peak/rms loudness levels, as well as spectrum and history
    c. Options to focus on individual tracks.
    d. Highly technical needs additional knowledge of mastering techniques.
2. **Voxengo SPAN with K-System Support**
    a. A spectral analyser and loudness metering tool
    b. Can be set to K-14 loudness standard, to give indication of more headroom.
    c. No traffic light colour display for loudness metering like Izotope

   d. Simple product that is still a bit complicated for beginner musicians.

3. **iZotope Neutron Pro with Mix Assistant**
   a. Easy-to-use mixing assistant that helps users create a more balance mix through balancing the frequency spectrum content more closely to set genre standards, or even a custom reference track. However, it doesn't provide a visual display of what's happening dynamically in the overall mix, just provides ways to maximise balance which impacts optimum dynamics of the track. Adjustable elements of the plugin are key areas that help the overall perception of loudness and dynamics, so looking into these features may prove useful too.

Roadmap and Milestones

As a solo developer, you'll need to focus on the most critical aspects of plugin development to create efficient milestones. Here's what to prioritize:

Practical Milestones

Milestone 1: Working Prototype (2-4 weeks)

- [ ] Core DSP algorithm functioning
- [ ] Basic parameter control
- [ ] Loads in at least one DAW
- [ ] No crashes during basic operation

Milestone 2: Feature Complete (4-8 weeks)

- [ ] All planned DSP features implemented
- [ ] Functional UI with all parameters accessible
- [ ] Cross-DAW compatibility
- [ ] Proper preset handling

Milestone 3: Release Candidate (2-4 weeks)

- [ ] Refined UI
- [ ] Performance optimization
- [ ] Basic documentation
- [ ] Small set of useful presets

- [ ] Stability testing

Time-Saving Tips

1. **Use frameworks** like JUCE to handle cross-platform compatibility
2. **Limit scope** - a focused plugin that does one thing well is better than feature creep
3. **Leverage templates** for UI components rather than building everything custom
4. **Test with real projects** early to ensure practical usability
5. **Automate testing** where possible to catch regressions quickly

Must-Focus Areas

1. Core DSP Functionality First

- [ ] Build the essential audio processing algorithm that defines your plugin
- [ ] Start with a minimal working implementation before optimizing
- [ ] This is your product's foundation - get it right early

2. Minimal Viable UI

- [ ] Create a simple, functional interface just to control key parameters
- [ ] Don't get caught in design details initially
- [ ] Focus on parameter mapping and real-time control

3. Regular Testing Across DAWs

- [ ] Test frequently in 2-3 major DAWs you're targeting
- [ ] Catch compatibility issues early when they're easier to fix
- [ ] Validate core functionality works consistently

Remember that your first plugin doesn't need to be perfect. Getting something functional and releasing it will teach you more than endlessly refining in isolation.

Images

Knowledge gaps

☑ **Knowledge Gaps:**

- Optimal audio listening levels for both enjoyment and technical quality may be subjective, will require user feedback
- To consider the dynamic nature of individual listening preferences, and genre, may be subjective, will require user feedback
- Will producing music and audio with maximised dynamic range reduce ear fatigue? Will this increase the length listening and producing sessions? may need user feedback
- Considering the Fletcher-Munson curve as a function of time, may be required.
- Consider the effect of saturation effects on perceived audio levels and listening enjoyment and sense of 'live-ness' of the audio. What's happening here? I've felt much more enjoyment when using saturation plugins on the whole music creation process in general. Is it something that is mimicking more closely the way audio is reproduced in nature. What is wrong with the digital signal, that an analogue signal makes more pleasant?

☑ Could an MVP be made using a technology you already know how to use?

Yes, I may be able to create a working proto-type with effects plugins like FL Studio Patcher. I need to know the maths behind the fletcher-munson curve and current dynamics loudness metering plugins to see if I can produce a display which gives a read -out correctly

☑ Are there any advanced maths or science that I need to learn or understand in order to build?

The are yes. The Fletcher Munson curves are a set of experimentally determined graphs that show how loud—in dB SPL—a sound at one frequency must be in order to be perceived as equally loud as a sound at another frequency. This impacts how individual frequencies are perceived in volume, at the same SPL (Sound Pressure Level). The human ear is not a fixed listening device, and is dynamic in many ways. There may also be a temporal change in how certain frequencies are heard over time, and not just in this fixed time measurement. For example, when exiting the music venue, your ears sound 'muffled' and sounds outside sound dampened. This usually resolves itself. I believe this effect may be happening on a shorter time and smaller volume spectrum, and integral into why we have different emotional responses to types of music, and sounds in general.