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THE AUTONOMOUS POST PRODUCTION OF A PIANO RECORDING

Supervisor: Dr. Ian S. Gibson

This investigation aims to determine if the basic post production techniques typically applied to a piano recording can be done so entirely autonomously by software, in the form on a VST Plugin. Using spectral analysis the software will compare the incoming audio to a predetermined ideal and apply compression and equalisation accordingly, then alter the parameters of the effects in real time in order to maintain a relatively constant tone and volume. In the context of a popular music production this will be extremely useful during the mixing process, as it will automatically control the large dynamic and frequency range of the piano.

- The tonal qualities of a **time** domain / digital signal are determined via **frequency** domain analysis.
- **Conversion** between the time and frequency **domains** is achieved with the **Fourier** Transformation.
- The Discrete Fourier Transform (DFT) calculates the Fourier Transform of a **discrete** signal.

$$X_{k} = \sum_{n=0}^{N-1} x_{n} e^{-\frac{2\pi j}{N}kn} \qquad k = 0, \dots, N-1$$

• The Fast Fourier Transform (FFT) is an algorithm for efficient computation of the DFT.

Spectral Analysis



• Observing the **frequency content** of a signal in this way is called **Spectral Analysis**, and shows the amplitude of component sine waves in the signal.



- cal instruments.
- shape.
- Newton algorithm.



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• Besides the combination of signal analysis and effects processing, the merging of



compression and equalisation (EQ) is the main unique quality of the plugin.

• Compression-like **envelope following tech**niques are used to control the gain of parametric filters.

This results in a capability to perform **both** compression and compansion, as the filter gain can be positive or negative.

Processing

• **Filter gain** is determined by the difference in the **Y-axis** between input and ideal trace.

• Filter centre frequency and Q factor is deter-



mined by differences in the X-axis between traces.

• A high pass filter will also be implemented, who's cut-off frequency is determined by the first point where the traces cross. • These processes will result in a **piano tone** extremely **difficult to create** with existing products.