Dynamics

1 Introduction

- Sound pressure level of natural sounds varies over time
  - For example in music and in movies, the loudness of sounds works as an effect and is part of the content
  - When recording, the distance to microphone may vary, which causes unwanted level variation
- When audio is played back, the useful dynamic range may be limited due to background noise for example

Applications of dynamic range control

- **When recording**, it is desirable to use the full amplitude range optimally
  - Also to protect AD converters from overloading (clipping)
- Noise gates are used to suppress low-amplitude noise
  - Only audio signals exceeding a certain level will be passed through
- In record production, dynamic range variation can be limited at will
- In various recording formats, the limited range of amplitudes should be optimally used
- **When playing back** music or speech e.g. in a car, dynamic range variation has to be matched to background noise
  - Listening becomes easier / possible

Figure: block diagram of dynamic range control

1. Measure the level of input signal
2. Multiply the delayed input signal by factor $g(n)$
   \[ y(n) = g(n) \cdot x(n-D) \]

- Delaying signal $x(n)$ compared to the control signal $g(n)$ allows "predictive" level control (gain goes down before any "big bangs")
- Value of gain factor $g(n)$ is calculated in two steps
  - Static curve defines desired output level corresponding to the input level
  - Temporal variation of $g(n)$ is smoothed using a lowpass filter

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2 Static curve

- Figure: relationship between input level and weighting level is determined by a static level curve $G[\text{dB}] = f(\ X[\text{dB}] )$
  - Output level and weighting level are functions of input level
  - Thresholds: $LT$=limiter threshold, $CT$=compressor thr., $ET$=expander thr., $NT$=noise gate thr.

2.1 Operation regions

- **Limiter** limits the output level when the input level exceeds the limiter threshold $LT$
  - All input levels above the threshold lead to a constant output level
- **Compressor** maps a change of input level to a smaller change in the output level
- **Expander** increases changes in the input level to larger changes in the output level
- **Noise gate** is used to suppress low-level signals and to reduce noise

Threshold values in different parts of the static curve determine the lower limit for limiter and compressor and upper limit for expander and noise gate.

2.2 Compression ratio

- **Compression ratio** is visible in the logarithmic representation of the static curve: ratio of input level change $\Delta P_I$ to output level change $\Delta P_O$
  \[ R = \frac{\Delta P_I}{\Delta P_O} \]

- Typical compression ratios: $R = \infty$ (limiter), $R > 1$ (compressor), $0 < R < 1$ (expander), $R = 0$ (noise gate at threshold)

From figure: line equation $Y = CT + \left( \frac{1}{R} \right) \cdot (X - CT)$ and compression ratio $R = (X - CT)/(Y - CT)$

Switching from logarithmic to linear representation we obtain

\[ R = \frac{\log_{10}(x/c_T)}{\log_{10}(y/c_T)} \]

where $x$ and $y$ are linear levels, $c_T$ is linear compression threshold

From here we can solve linear output level $y$ as a function of the input level $x$

\[ y/c_T = 10^{\frac{1}{R} \log_{10}(x/c_T)} = \left( \frac{x}{c_T} \right)^{1/R} \]

\[ y = c_T \cdot \left( \frac{x}{c_T} \right)^{1/R} \]

The control factor $g(n)$ can be calculated as

\[ g(n) = \frac{y}{x} = \left( \frac{x}{c_T} \right)^{1/R-1} \]

\[ g(n) = \frac{y}{x} = \left( \frac{x}{c_T} \right)^{1/R-1} \]
3 Dynamic behaviour

- Besides static curve based level control, the dynamic (time-varying) behaviour of the control factor (attack and release time) plays a significant role in sound quality.
- The rapidity of input signal level measurement using PEAK or RMS algorithms also affects the rapidity of the overall dynamic range control.
- In the following, we first consider the level measurement and then the control of attack and release times by filtering control factor values over time.

### Level measurement

- **Output of the PEAK algorithm for a rectangular pulse**

![Graph showing the output of the PEAK algorithm for a rectangular pulse.](image)

#### PEAK algorithm (produces a level measurement $x_{\text{peak}}(n)$):

- Coefficient $AT$ determines the attack time, coefficient $RT$ the release time.
  
  - If $|x(n)| > x_{\text{peak}}(n-1)$
  
  $x_{\text{peak}}(n) = (1 - AT) x_{\text{peak}}(n-1) + AT \cdot |x(n)|$
  
  - Else
  
  $x_{\text{peak}}(n) = (1 - RT) x_{\text{peak}}(n-1)$

  - $AT = 1 - \exp(-2.2 T_s / (t_a/1000))$, where $t_a$ = attack time (ms), $T_s = 1/F_s$ is samp. interval.
  
  - $RT = 1 - \exp(-2.2 T_s / (t_r/1000))$, where $t_r$ = release time (ms).

  - Typically $t_a = 0.02 \text{ms} ... 10 \text{ms}$ and $t_r = 1 \text{ms} ... 5000 \text{ms}$.

- **RMS algorithm (produces level measurement $x_{\text{rms}}(n)$):**
  
  - Square of the input, averaging with first-order lowpass filter.
  
  - Temporal length of the averaging is determined by coefficient $TAV$.

  $x_{\text{rms}}^2(n) = (1 - TAV) \cdot x_{\text{rms}}^2(n-1) + TAV \cdot |x(n)|^2$

  - $TAV = 1 - \exp(-2.2 T_s / (t_M/1000))$, where $t_M$ = averaging time in milliseconds.

  - In the figure $t_M = 100 \text{ms}$ (green line).

  - For peak (red line): on the left, $t_a = 10 \text{ms}$, on the right $t_a = 0.02 \text{ms}$.

- **Level measurement**

![Graph showing level measurement over time.](image)
3.2 Gain factor smoothing

- Attack and release times can also be implemented by smoothing the temporal variation of the control factor $g(n)$

1. Control factor $f(n)$ is compared to its previous value (control factor, calculated using the static curve, was denoted by $g(n)$ above)
   - Determine whether the control factor is in attack or release state
   - Accordingly, assign variable $k$ either value AT or RT

2. Value of the control factor is obtained by filtering the control factor values with first-order lowpass filter:
   
   $$g(n) = (1 - k) \cdot g(n - 1) + k \cdot f(n)$$

4. Implementation

4.1 Example: limiter

- Figure: limiter block diagram
  1. Level $x_{\text{peak}}(n)$ of input signal $x(n)$ is measured using the PEAK algorithm
  2. Value $\log_2[x_{\text{peak}}(n)]$ is compared to limiter threshold $LT$
  3. If the value is above the threshold (difference is positive)
     - Difference is multiplied by the negative slope $-LS$ of the limiter
     - Take antilogarithm $2^G$
     - The resulting control factor $f(n)$ is smoothed using first-order lowpass filter SMOOTH
  4. If the value is not above the threshold, factor $f(n)$ is set to value 1.
  5. Delayed input signal $x(n - D_1)$ is multiplied with the smoothed control factor $g(n)$ to give the output $y(n)$

4.2 Combination system

- The basic structure of compressor, expander, and noise gate is similar to the limiter
- Level measurement can be made using RMS-based value $0.5 \cdot \log_2[x_{\text{rms}}(n)]$
- Practical implementation is a cascade, where each part implement one of the basic operations (limiter / compressor / expander)
  - Appropriate parameters can be chosen for each stage
  - Because the maximum output level is reduced by the limiter, the static curve in its entirety can be shifted up with constant gain
- Figure: static curve of the combination system (additional static gain 10 dB)
Combination system: example

- It is seen that signals with high amplitude are compressed and those with low amplitudes are expanded
  - An additional gain of 12 dB (control factor maximum value is 4)
  - \( g(n) = 4 \): operating in the linear region of the static curve
  - \( 1 < g(n) < 4 \): compressing
    - Achieving a louder output signal overall
  - \( 0 < g(n) < 1 \): expanding
    - Dynamic range of low-amplitude signals is increased
  - Operation of the noise gate is not visible in the example

- Note: the control factor indeed decreases the level of the expanded part of the signal (if the additional gain of +12 dB is not taken into account)
  - That way compression ratio \( R = \Delta P_{\text{in}} / \Delta P_{\text{out}} \) becomes larger than 1 and level changes in input are mapped to larger changes in output
  - Visible in the static curve of \( G \ [\text{dB}] \) in §4.2

4.3 Computational aspects

- The sampling rate of the level measurements \( x_{\text{peak}}(n) \) and \( x_{\text{rms}}(n) \) can be reduced by factor four
  - Because the signals are by their nature already bandlimited, we can simply pick every fourth value (without aliasing)

- The computation can be spread / shared as follows:
  - Levels PEAK/RMS are updated for each input sample
  - The following modules are executed one at a time every four cycles: (1) LD(x): logarithm, (2) CURVE: static curve, (3) \( 2^n \): antilogarithm, (4) SMO : control factor smoothing
  - Values of the control factor are interpolated (upsampled) by repeating each value 4 times and by multiplying (MULT) the input signal to the output

5 Stereo signals

- For stereo signals, a common control factor \( g(n) \) is needed
  - That way the stereo balance is not displaced
  - Input signal level is measured from channel average

- Figure: dynamic range control of a stereo signal

6 Bandwise dynamic range control

- When processing complex audio signals, dynamic range control is sometimes performed bandwise
- For example 4-band filterbank followed by compression at each band separately
- De-esser: dynamic range control at sibilants (/s/ phoneme) frequency range (about 5 kHz)
  - Bandpass filter at 2-6 kHz is used to detect prominent phonemes.
  - If a threshold value is crossed, a so-called peak filter is used to cut the level of /s/ phonemes in the same frequency range (peak filter: has unity response everywhere else except around its center frequency, where either boost or cut can be applied)