CADENZA Software

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NOTICES

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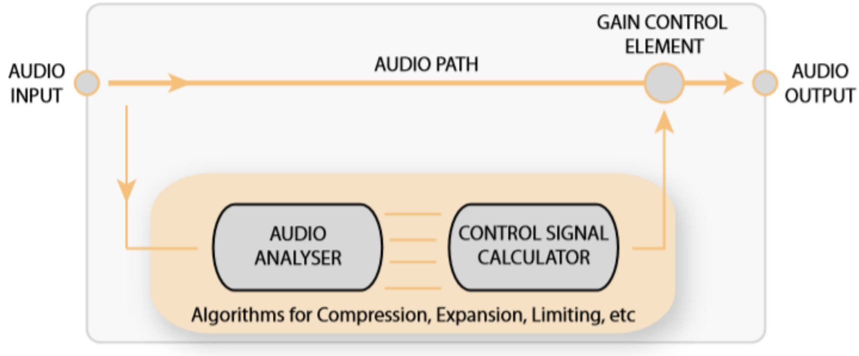
1 Introduction

CADENZA is a complete dynamic range controller for digital audio passing from input to output, providing the user with **Compressor**, **Limiter**, and **Expander** functions.



The Compressor action is adapted to human hearing instead of being a purely technical gain servo system. The parameters preserve the original impression of dynamics and life of the sound: Beat music still has a driving beat at the compressor output.

Presence and stability of the sound scene is maintained as the Compressor is not irresolutely compensating every minor level difference. A control signal cleaning function further reduces distortion normally introduced by fast compression or limiting.

Due to the prevalence of IP based audio distribution, the CADENZA software is optimised for a server-based system architecture supporting high-density AES67 audio inputs.

The CADENZA software is controlled through an intuitive web-based GUI and performs the following functionality:

- Load previous/default configuration from a configuration file (Presets).
- View audio input and output levels.
- View the current gain being applied to any audio input in the form of a coloured bar display.
- View the input/output relationship based on the current audio process parameters in graphical form.

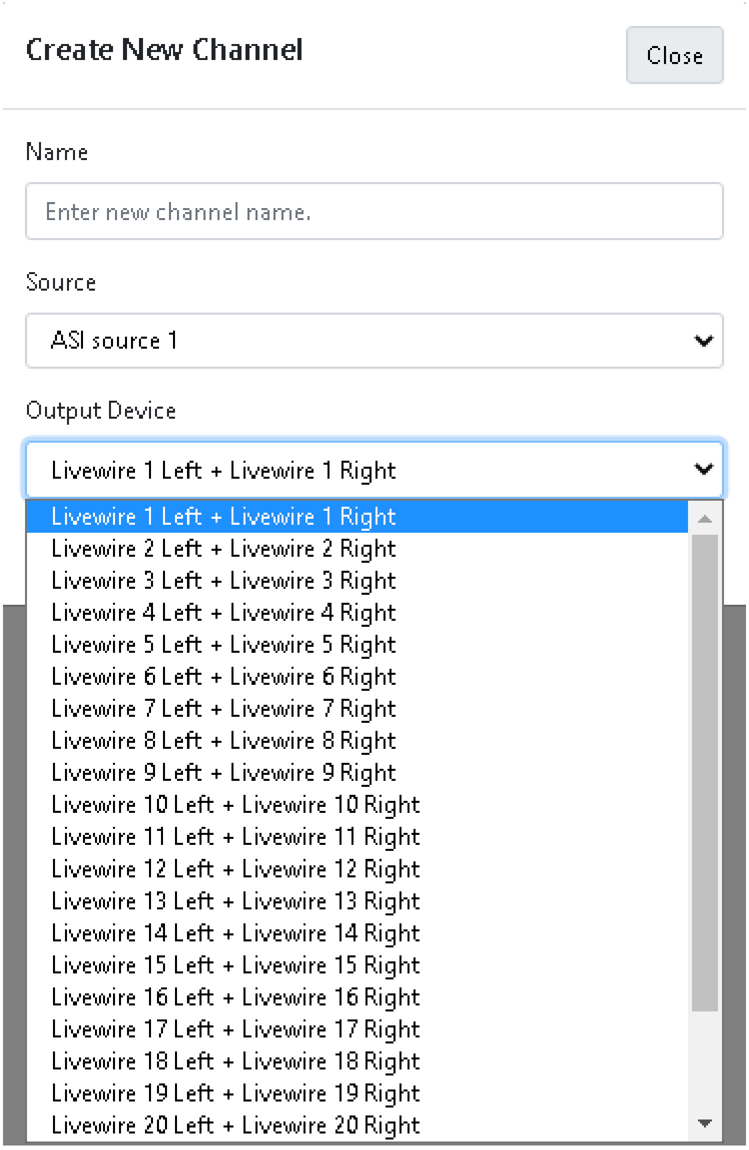
This operational guide documents the general functionality and more in-depth technical descriptions of the audio process.

2 Graphical User Interface

The CADENZA software GUI provides an overview of multiple channels by user-assigned name. Expanding each channel accesses a comprehensive dashboard allowing the user to set all parameters controlling the operation.

2.1 Channels

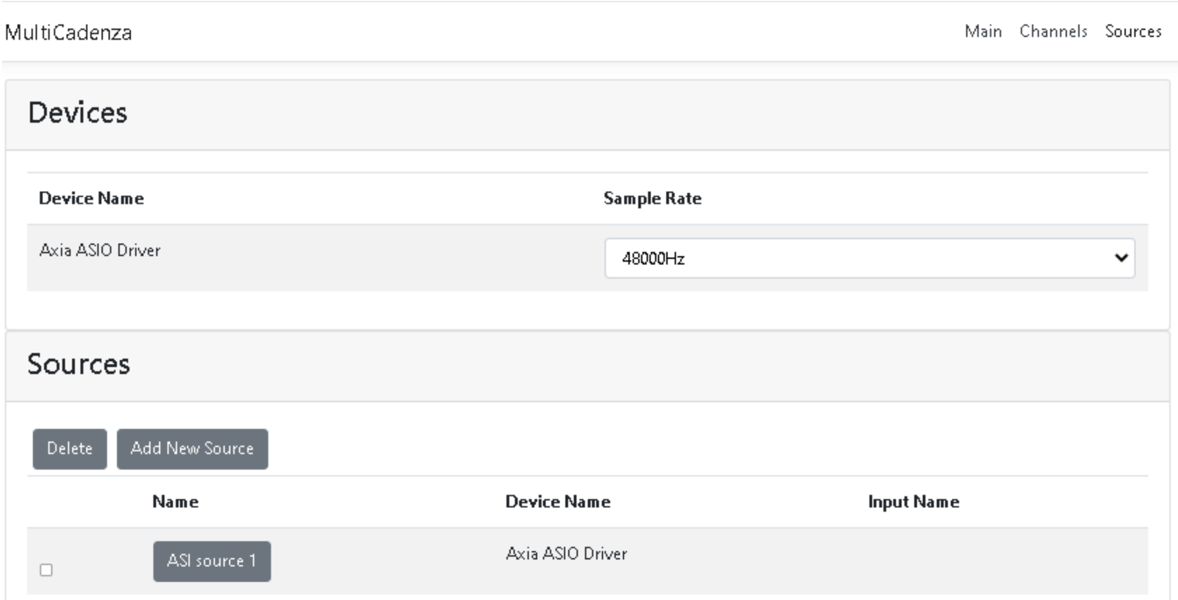
The format of the present input audio is shown in the **Channels** tab, displaying some key properties of the incoming audio. The user may designate a new channel, select source, and output device.



2.2 Sources

CADENZA will present the device name, and indicate actual sample rate of input audio:

- 32k = 32kHz sample rate
- 44k = 44.1kHz sample rate
- 48k = 48kHz sample rate



The user may create and delete sources associated with the recognised device under the **Sources** tab.

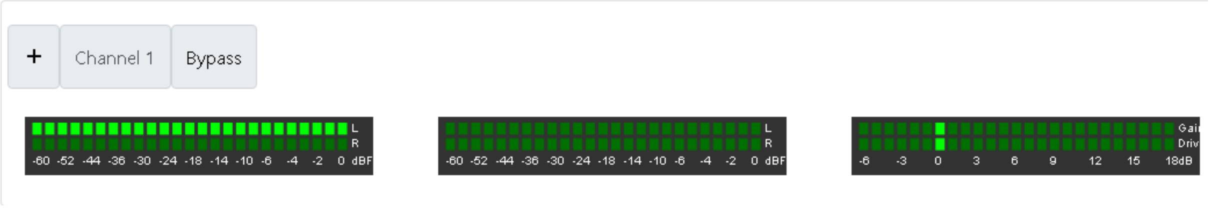
2.2.1 Deviation from nominal sampling frequency

The deviation is classified in two intervals:

- <400 ppm = frequency deviation less than 400 ppm
- <4% = frequency deviation less than 4%, but worse than 400 ppm

3 Main (dashboard)

The dashboard displays channels by name, and level meters to show the real time processing of the audio.



1 **2** **3**

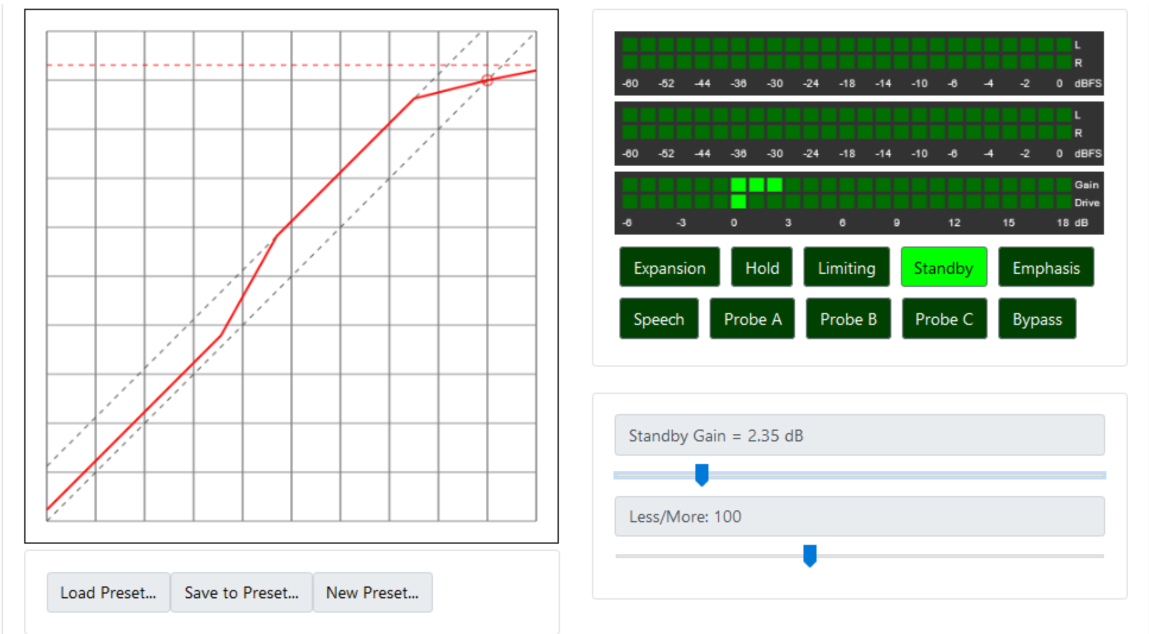
3.1 Level Meters

Level Meters show absolute peak readings (single sample). The peaks are held for about 10 ms and then released with a time constant of about 0.1 s.

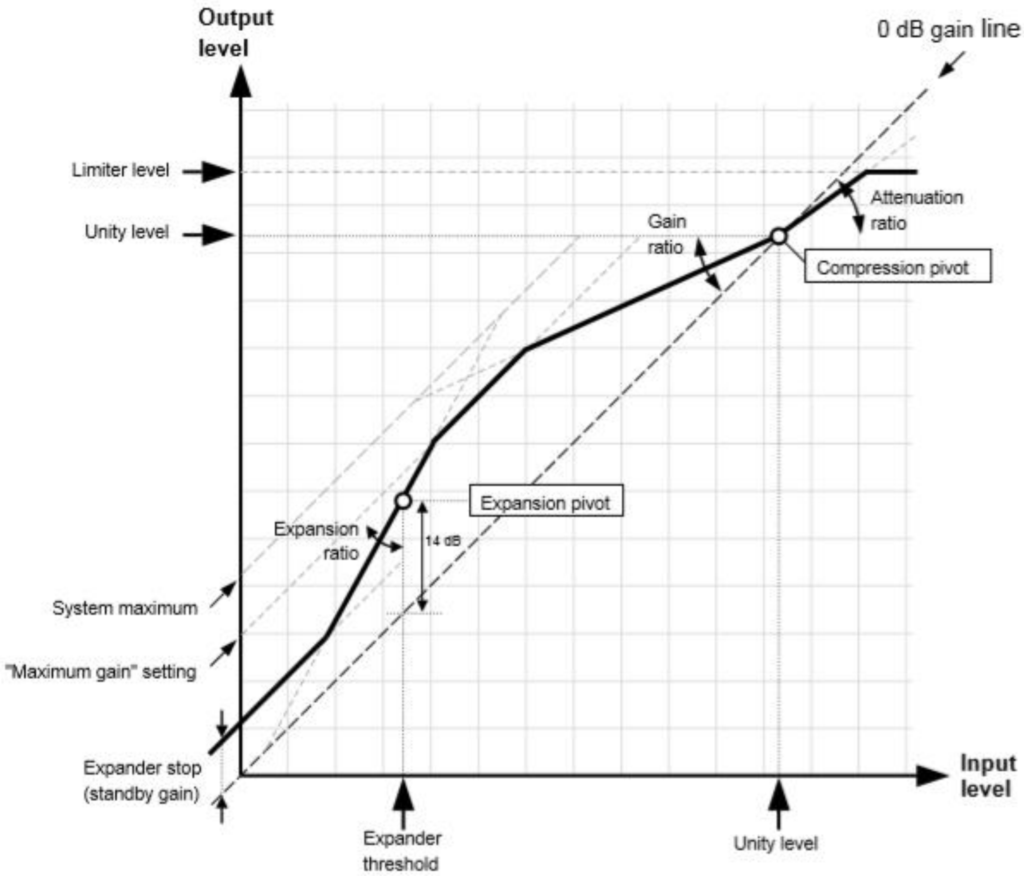
1. **Input Level**: indicates that the input audio level (L & R).
2. **Output Level**: indicates that the output audio level (L & R).
3. **Gain Meter**: The upper bar displays the **momentary gain** for the processor. Each LED segment represents 1 dB gain reduction. The lower bar displays the **static gain** contribution to the total gain.

3.2 Expanded Channel

Each channel can be expanded to access the control panel for user configuration.

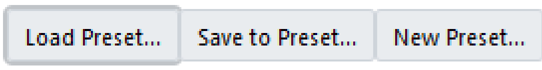


3.2.1 Gain Control Diagram



The Gain Control graph dynamically adjusts in relation to the changes in configuration from the control panel.

3.2.2 Presets

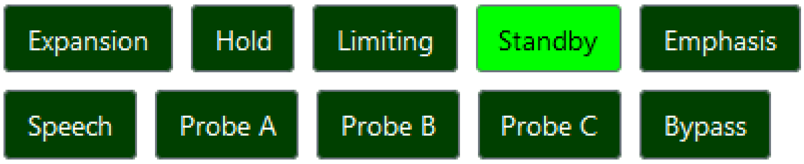


The audio process parameters are organised in presets containing values and settings that control the CADENZA algorithms.

The dashboard provides three functions for presets:

1. **Load Preset** allows for configurations to be imported and populate the control panel parameters.
2. **Save to Preset** allows the current configuration to be saved, overwriting the initial loaded preset.
3. **New Preset** allows for a new configuration to be saved to the Preset library.

3.2.3 Function Indicators



By means of virtual LEDs, the user can monitor the state of some essential functions:

3.2.3.1 Expansion

LED indicating that the Expander is in command of the gain. **Expansion** and **Hold** lit simultaneously indicates that the gain changing has stopped, and that the Expander will start reducing the gain when the user-controllable Expander *Delay* period has expired.

3.2.3.2 Hold

LED indicating that the Compressor/Expander has entered the *Hold* mode, which will stop the gain from increasing.

3.2.3.3 Limiting

LED indicating that the audio level at the compressor output has exceeded the *Limiter Level* setting, and that the Limiter is active.

> The Limiter gain reduction is part of the gain meter reading, however limiting might occur on the gain side as well as on the attenuation side depending on the compressor gain contribution.

3.2.3.4 Standby

LED indicating the input level is below the global *Standby Level* setting.

3.2.3.5 Emphasis

LED indicates the *Emphasis* filter is active.

3.2.3.6 Speech

LED indicating that the audio process parameter settings for speech are selected.

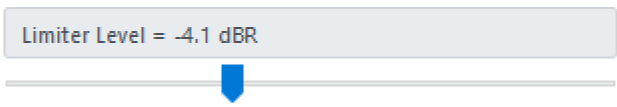
3.2.3.1 Probe A, B, C

LED indicating that the selected logic condition for the function monitor designated “Probe A/B/C” is true.

3.2.3.2 Bypass

Pressing the Bypass button toggles between bypassed and engaged mode of operation. In bypassed mode, the audio is not processed by the audio processor. Please note that any static gain or attenuation added when level aligning the unit will also be bypassed. In the bypass mode the audio path is fully transparent. The **Bypass** LED indicates the bypass mode is active.

3.2.4 Slider Control

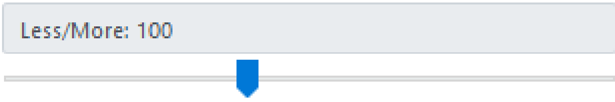


Control options which do not have a drop-list can be adjusted by selecting a parameter and sliding the dial below the level meters to the wanted value.

The selected parameter will highlight red to indicate it is being adjusted.

The slider automatically sets up limits and steps as per the parameter being configured.

3.2.5 Less/More Control



The first level of user control of the audio processing is operating the **Less/More** control. By means of one single slider control the user may scale the effects of the presently selected type of processing. The default settings are 100%, and the effects may be scaled from 0% to 150%. The attenuating and limiting functions are still active at 0%.

0% setting: No gain is possible. However, the programmed limiting and compressor attenuation will be performed.

10% to 150% setting: The percentage is a subjective comparison to the original power of the currently selected preset. Low audio input levels will be amplified according to the selected percentage. Also, the release speed will be affected accordingly.

3.2.5.1 Adjusting the Less/More control

Scenario 1: The difference between loud and quiet parts of the programme has become too small after the processing (the dynamics are too reduced). Everything is equally loud. It might also be like the quiet parts sound *closer* than the originally louder parts, a zoom effect.

Solution: Lower the **Less/More** setting.

Scenario 2: The over-all sound is tame. The quiet parts are difficult to hear (the dynamics are not reduced enough).

Solution: Heighten the **Less/More** setting.

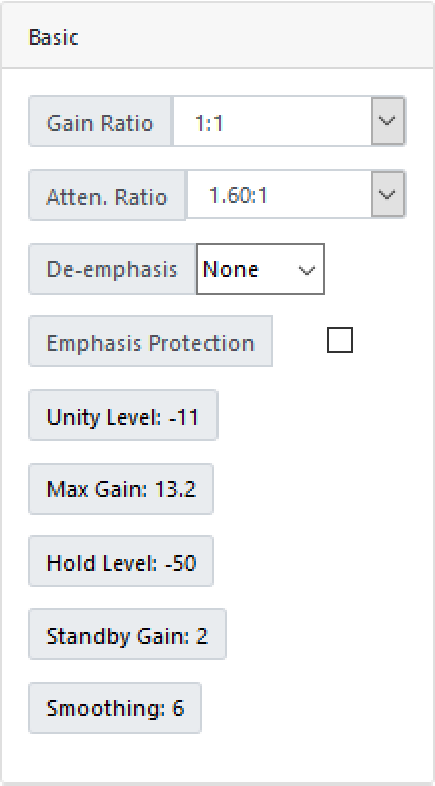
Consider the **Less/More** control as a volume control affecting only the quiet pieces of audio. With this control you can gain-up the quiet pieces without gaining-up the loud ones.

High input levels will normally stay almost unaffected, close to 0 dB gain. This level/gain area is like a stationary pivot of the process system. The very loud peaks will be attenuated in the audio path processor.

The gain meters will show the gain as a deflection to the right of the 0 dB segment. When lowering the **Less/More** setting the gain activity is contracting towards the 0 dB segment. The audio path gain meter may show occasional attenuation at loud peaks. Attenuation is displayed by a meter deflection to the left of the 0 dB segment.

The set *Standby Gain*, which is the start gain after a non-modulation period, will be scaled in accordance with the **Less/More** percentage.

3.3 Basic Controls



In the **Basic** menu, important parameters can be changed when adapting a preset, to optimise the audio process performance according to the specific needs for the actual service. Most Basic parameters define the compression diagram.

3.3.1 Gain Ratio

The compression ratio used when the compressor is amplifying. That is for input levels below the *Unity level*. The compressor's activity in this area is displayed by a gain meter deflection to the right of the 0dB segment.

3.3.2 Attenuation Ratio

The compression ratio used when the compressor is attenuating. That is for input levels above the *Unity level*. The compressor's activity in this area is displayed by a gain meter deflection to the left of the 0dB segment.

3.3.3 De-emphasis

The *De-emphasis* options from the drop list are:

- **None**: *De-emphasis* will not be applied. Any emphasis flag at the input will be cleared before sent to output.
- **J.17**: *De-emphasis* EQ and phase shift according to the CCITT J17 norm will be applied, regardless of signalled emphasis at the input. The applied J17 de-emphasis has the 0 dB point @ 0 Hz. The emphasis flags of the audio output stream will be set to "no pre-emphasis".
- **50us/15us**: *De-emphasis* EQ and phase shift according to the 50/15µs norm will be applied, regardless of signalled emphasis at the input. The applied 50/15µs de-emphasis has the 0 dB

point @ 0 Hz. The emphasis flags of the audio output stream will be set to ”no pre-emphasis”.

3.3.4 Emphasis Protection

Selecting this function adds in some compensation for any downstream equipment that has pre-emphasis applied. Pre-emphasis is where the equipment applies high-frequency gain ("treble boost") to improve the signal-to noise ratio:

- Noise (hiss) is more noticeable at higher audio frequencies (hence why we hear a hiss rather than a rumble).
- Pre-emphasis boosts the high frequency content of the signal before transmission.
- At the receiving end we have the original signal plus any noise introduced in the transmission process.
- De-emphasis is applied to reverse the effect of the pre-emphasis, i.e., by cutting the high-frequency content of the signal.
- This also cuts down the high-frequency noise, so overall it sounds better.

Setting the *Emphasis Protection* on the Cadenza allows it to take this process into account, so it does not over-boost the signal and cause overloading in the downstream equipment.

3.3.5 Unity Level

Defines the fixed point of the compressor. A constant input level corresponding to the *Unity Level* setting will have 0 dB gain in the compressor, regardless of other parameter settings. This is the compressor’s pivot. Lower input levels may be amplified, and higher levels may be attenuated (other parameter settings determine whether the dynamic operation will be performed or not).

3.3.6 Max Gain

The absolute limit for the audio process gain. Regardless of other parameter settings, the total dynamic gain can never exceed the *Maximum Gain* setting. This is displayed by the gain meter.

3.3.7 Hold Level

An input level-controlled gain freezer. Immediately when the input level is below the set *Hold Level*, the compressor will be locked at the present gain value. The LED marked **Hold** will light when the gain is locked.

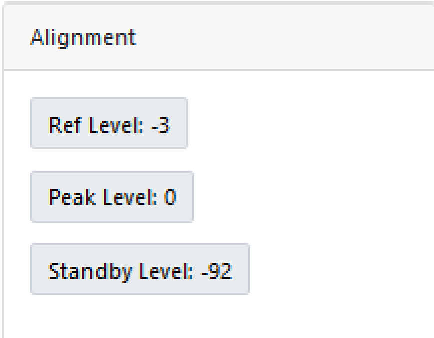
3.3.8 Standby Gain

The *Standby Gain* setting defines the final gain for an expander fade, and the start gain when the processing becomes active again. The *Standby Gain* is scaled in accordance with the **Less/More** setting.

3.3.9 Smoothing

Smoothing is part of the final control signal cleaning algorithm. *Smoothing* may remove the small, rapid changes and sharp peaks of the raw control signal. The large rapid changes are not affected. No time constant is introduced by the control smoothing function (it is not a filter). This function further increases the stability of the compression. A high setting may introduce hole punching after loud sounds.

3.4 Alignment Controls



The **Alignment** parameters are global system parameters, valid for all audio process presets.

- *Ref Level* may be set to the average peak level of the input audio.
- *Peak Level* may be set to the desired maximum peak output level.
- *Standby Level* may be set to any level from –110 dBFS to –60 dBFS.

3.4.1 Reference Level

"Tuning" to the correct setting by using the *Ref Level* control as an input gain control by observing some of the effects described below:

Lowering the *Ref Level* means gaining up.

Indication	Action to take
The **Limiting** LED does not light.	Lower the Ref Level.
The **Limiting** LED is periodically, nearly constantly lighting.	Heighten the Ref Level.
The **Limiting** LED is blinking on the very loudest parts.	Correct Ref Level setting.
The **Gain Meter** show several dB gains, although the input level is at its loudest.	Lower the Ref Level.
The **Gain Meter** shows constant attenuation, although the input level is not at its loudest.	Heighten the Ref Level.
The **Gain Meter** is close to the zero segment when the input level is at its loudest.	Correct Ref Level setting.

3.4.2 Peak Level

The user defines a maximum peak output level, while the CADENZA software automatically adds the appropriate static gain or attenuation.

The automatic level alignment will adapt the processed audio to the set peak output level. Consequently, each time the **Limiting** LED lights, some peak at the set *Peak Level* will be present.

The default setting for the maximum possible output peak level is 0dBfs.

3.4.3 Standby Level

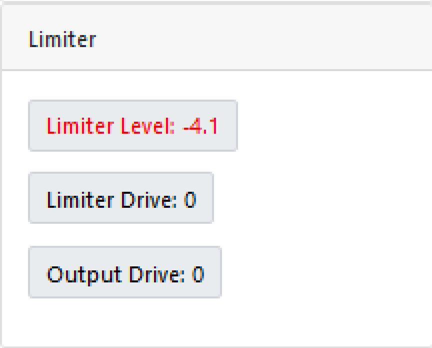
When the audio input level is below the *Standby Level* setting, the standby mode is entered. The idea is to put some essential process parameters in "get ready" positions when there is no modulation at the audio input. This is mostly for avoiding great gain changes or excessive gain at the moment when modulation returns.

Standby mode features When entering the standby mode, this will happen:

- Function indication: **Standby** LED is lit.
- Gain switch: The audio processor gain is switched to the *Standby Gain* setting.
- Attack calculator reset: The attack speed calculator data inputs are updated for the “Get ready” position.
- Release switch: The main release speeds switches to the value that is valid for the actual gain in the processor.
- Process mode and speech detector reset: If the current audio process preset has different music and speech modes, the favoured mode is selected. If the speech detector is used for the mode controlling, it is set to the “start search” position.

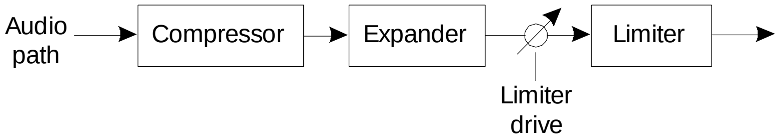
The default setting is –82 dBFS. When there is a control desk feeding the system, the standby mode should be entered when all faders are pulled down (closed), or at digital silence. Once this level is found, heighten (make it less negative) the setting some 3– 5 dB to get a distinct switch function. Mind the difference between the enter and exit levels. When the idle noise is below some –85 dBFS an exact match is not very important.

3.5 Limiter Controls



The **Limiter** settings are crucial for an optimised audio output level. If the audio output from the compressor does not reach the *Limiter Level*, the output peaks will not be at the desired and set *Peak Level*. This is because the *Limiter Level*, will be transformed to the user defined *Peak Level* by an automatic function in the *Alignment* procedure.

The processor layout for Limiter with *Limiter Drive* is as follows:



3.5.1 Limiter Level

The *Limiter Level* is a **dBr** parameter. The Limiter action prevents this level threshold from being exceeded. If the amplitude of the peak is less than or equal to the *Limiter Level*, it will slip through totally unaffected. If the amplitude of the peak is larger than the limiter level, it will be reduced to be equal to the limiter level. Not more, not less, no overreaction, no clipping.

The limiter action is not just superimposed on the compressor action, which would make an overreaction possible. Instead, the limiter and the compressor co-operate to prevent any counteraction.

3.5.2 Limiter Drive

Using the gain potentiometer between the Compressor and the Limiter is the easy and intuitive way to adjust the Limiter action. If the Limiter action is too aggressive or too vague, watch the **Limiting** LED and adjust the potentiometer until the desired Limiter action is achieved.

As default in all presets the audio path processor's *Limiter Drive* has the 0.0 dB setting. The limiter action is factory adjusted by means of the *Limiter Level*. The two controls can very well be used in combination.

If the ***Limiter Drive*** has already been used to set the appropriate limiter action, do not alter the ***Limiter Level***.

3.5.3 Output Drive

This is a gain potentiometer, used for controlling the loudness at the audio output. When the *Output Drive* control is set at positive gain values, the Output Limiter will be engaged to take care of the extra peak levels. This method may be used for extra loudness increase. Normally, settings up to +2.5 dB or more can be used without introducing audible artefacts. One early artefact is sunken transients and loss of high frequency energy in the transients. Too high settings will result in audible wave form deformation (not clipping however) or limiter flutter.

Used moderately the result is often an improvement. Gaining up at the Output Limiter's input is like feeding a clipper. Each added 0.1 dB will give the corresponding higher average audio level.

3.5.3.1 Adjusting the Limiter parameters

Limiter attack and Look-ahead

As the Limiter must be capable of establishing a definite maximum amplitude limit, the attack must be immediate (attack time = 0 s). However, the gain reduction starts 2 ms before the momentary output amplitude from the compressor is about to cross the limiter level. With this *look-ahead* function the peaks are precision controlled without clipping being involved.

Limiter release

The Limiter release is another example of the unique co-operation between the Limiter and the Compressor. There is no special Limiter release. The Compressor's release intelligence is used also for the Limiter. This means that the Limiter function is not as dangerous to use as in other processor designs where a hard-working limiter will cause either peak riding flutter or level ducking followed by sound holes.

Limiter action monitoring

As an indication that the *Limiter Level* is reached, the **Limiting** LED is activated. This also tells that the limiter is active and that there are output peaks at the set *Peak Level*. Normally the **Limiting** LED should light up now and then on the loudest audio parts.

The amount of limiting is part of the gain meter's deflection and can be monitored as fast gain reductions. Because of the compressor's gain contribution, the limiting often occurs on the gain side of the 0 dB segment, where it may appear as fast contractions of the gain deflection.

Limiter activity is very hard to distinguish from Compressor activity on the gain meter as the same release functions are used. Furthermore, the human eye can hardly register a very short limiter action when it appears on the gain side of the meter. It is displayed though.

A preset may be partly based on the Limiter function in the audio path processor, making the **Limiting** LED light steady for long periods.

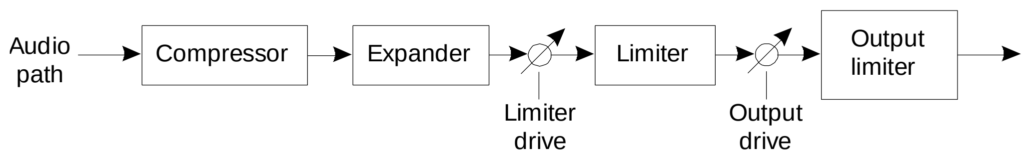
Average-to-absolute margin

A more normal setting is perhaps to have a little margin between the *Ref Level* and the absolute *Peak Level*. Set this margin with the *Limiter Level* control. At a +4 dBr setting there will be approximately 4 dB margin between the average peaks and the absolute peaks. The Compressor's attenuation ratio and attack parameters are very important for what will happen within this margin.

Moving the Limiter threshold up and down with the *Limiter Level* control, the automatic level alignment assures that the limited peaks will always lick the set *Peak Level*. Therefore, a lowering of the *Limiter Level* results in a louder sound. As the peaks are limited at a lower level the total audio will have the corresponding extra gain.

3.5.3.2 The Output Limiter

The final stage in the audio path is an *Output limiter*. This Limiter is working with 2 ms look-ahead and constant reaction times. Both the attack time and the release time are 2 ms.



Fully controlled limiting

The maximum possible gain reduction corresponds to the positive dB value set with the *Output Drive* control. When the processor performs a deeper limiting, the main Limiter takes care of the rest.

The CADENZA processing guarantees that no cascading phenomena or any other counteraction between the two Limiters occur. Instead, they support each other. When one of the Limiters cannot fulfil the present limiting task, the other Limiter will fill in with exactly the extra gain reduction needed.

Output limiter meter

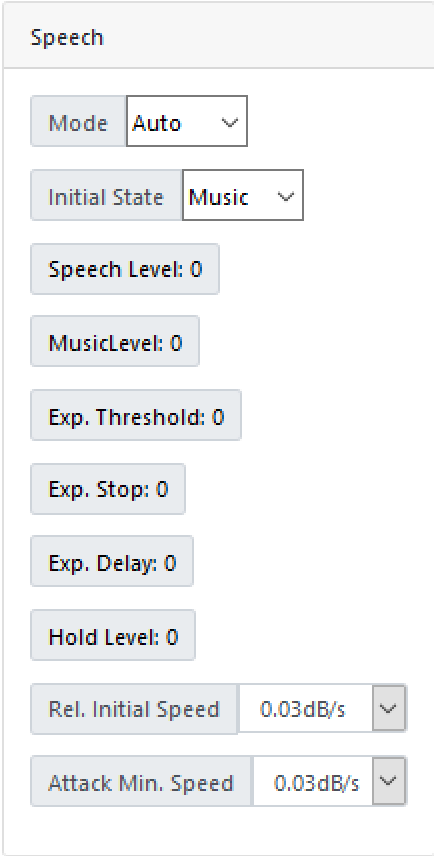
The audio path gain meter displays the output limiting. A bar emerging from the right indicates how many dB gain reduction the output limiter is performing. To make monitoring easy the meter deflection release has been slowed down.

Re-balancing the Limiters

In some presets, the Output Limiter has the main response for the limiting but is still supported by the main Limiter. A considerable output limiting can be allowed as it is set to occur during the Compressor's attack sequence only.

To accomplish such a re-balancing of the Limiter functions, the desired depth of output limiting is set as a positive gain at the *Output Drive* control, while the *Limiter Drive* setting is reduced with at least the same dB value. Go further down with the *Limiter Drive* until the desired balance is achieved.

3.6 Speech Controls



3.6.1 Mode

The options in the drop list are "Auto", "Speech" or "Music". With this selector in the **Speech** menu, differentiated music and speech processing can be decided upon.

3.6.2 Initial State

The options in the drop list are "Speech" or "Music". Some processor parameters can be set differently for music and speech. The LED **Speech** will light when the speech parameters are active.

3.6.3 Speech Level

By telling the detector in what level area speech can be found, the faulty detections can be reduced. Set the expected speech level with the *Speech Level* control, which is a dBr control (a relation to the *Ref Level*).

The level values correspond to the average level of the speech. For that reason, they are perhaps much lower than expected. For a classical channel with moderate levels at the announcements, the *Speech Level* could be as small as –20 dBr. For a rock or talk channel the *Speech Level* could be –9 dBr to –18 dBr. The default settings are –21 dBr for the classical and –12 dBr for all other presets.

3.6.4 Music Level

The detector can be adjusted to determine the level the music signal must exceed to be detected as music. This is also a long-term average level.

At higher *Music Level* settings, the detector will be more disposed for interpreting the audio signal as speech. The reverse is valid for lower settings. Too high a setting will result in much music being detected as speech. Not the least rock music, which very often has a more interruptive character than classical music. Naturally, rapping will very often be detected as speech. The instrumental background is the determining factor in this case.

If there is any rule, the *Music Level* should have a higher setting in a speech favouring preset compared to a music favouring one. A little lower than –30 dBr for speech favouring and a little lower than –40 dBr for music favouring. The default settings are in the –30 to – 45 dBr range.

3.6.5 Expander Threshold

The Expander function is normally involved to perform fades to normal gain at longer pauses. Start by enabling the Expander and adjust the *Expander Threshold* to the same level as the *Hold Level* or lower. Run dynamic talk material with and without background noise and observe the expander fades. If the expander starts fading at a too high input level, lower the threshold, and vice versa. Note that an eventual fade will stop at the set *Standby Gain* value. Below this gain will not action the Expander. Adjust the *Standby Gain* if necessary.

The **Expansion** LED indicates the expander take-over. An expansion sequence always starts with a gain hold period, indicated by both the **Expansion** and **Hold** LEDs lighting. The actual fade starts when only the **Expansion** LED is lit. The gain meter will display the fade.

3.6.6 Expansion Delay

After entering the expander mode, the fade-down will come to effect after the delay period set with the *Expansion Delay* control. The delay is inversely proportional to the difference between the audio input level and *Expander Threshold* setting. The larger the difference, the shorter delay. The delay is also proportional to the Expansion speed setting.

3.6.7 Hold Level

The general rule is to adjust the *Hold Level* to the lower end of the useful dynamic range of the input audio. In a talk channel, the *talk* is to be processed, not environmental sounds or studio atmosphere. When the *Hold Level* is properly set, the compressor will be alive only when somebody is talking. In the very short or longer pauses the hold function will be active to stop the gain from increasing. This gain stop has no relation to the actual compressor gain, only to the input level. It may occur anywhere on the gain scale.

The useful talk levels are perhaps as low as 40 dB below the *Ref Level* (= average peak level). Begin with *Hold Level* settings from –15 dBr to –40 dBr. Run talk material and watch the **Hold** LED. The LED should light when nobody is talking, also when there is background noise. The higher setting, the less active the processing, and more protection against background noise fishing.

Loud background noise may be caused if the *Hold* function is not engaged, amplifying noise in pauses. A higher *Hold Level* setting. Beware of very low frequency traffic sounds releasing this *gain brake* by using a bass cut or roll-off below 80Hz, at least for outdoor speech items.

When the correct *Hold Level* setting is found, a higher compression *Gain Ratio* and/or higher release speeds achieves dense talk processing without trash sounds being amplified in pauses.

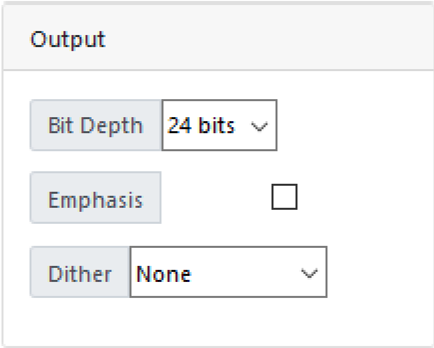
3.6.7.1 What parameters are switched?

There are no static gain changes involved in the parameter switches. Different speed and gain stop parameters are the most important changes. The compressor will use a higher minimum attack speed and a generally slower release for not favoured material, making the compression more relaxed at the same time as the integration time is shortened in the input audio weighting.

When the speech detector is active in the Talk/Mixed preset, a drama broadcast with much environmental sounds behind the dialogue will be classified as music. The release will slow down, reducing the risk of dialogue-controlled pumping of background sounds. The same goes for a sports broadcast from a public arena.

In presets favouring music, speech-like interruptive music will be detected as speech, reducing the pickup of background sounds between the musical bursts.

3.7 Output Controls

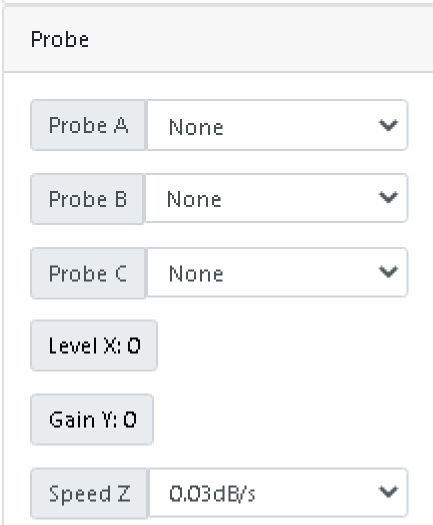


3.7.1 Dither

Dither is the noise applied to randomise the quantisation error. The *Dither* selector has three modes: *None*, *Triangular* or *Rectangular*:

- **None:** No dither noise will be used when quantisation is performed. This is the default setting.
- **Triangular**: white dither noise with a triangular probability density function (TPDF) will be used when quantisation is performed. The peak amplitude of the dither is $\pm$ 1 least significant bit.
- **Rectangular**: white dither noise with a rectangular probability density function (RPDF) will be used when quantisation is performed. The peak amplitude of the dither is $\pm$ 0.5 least significant bit.

3.8 Probe Controls



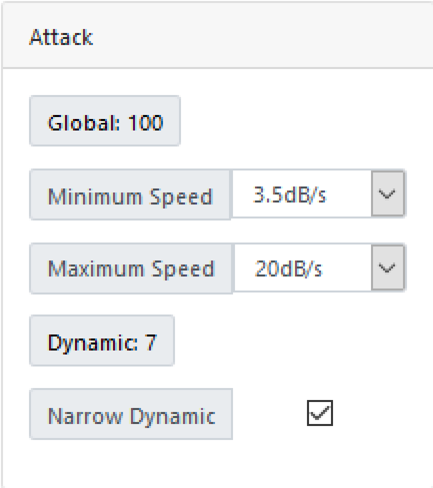
CADENZA features three **Probe** LEDs, respectively denoted *Probe A*, *Probe B* and *Probe C*.

The activation of these LEDs can be programmed to occur on specific audio input and processor action events. The purpose is solely for the user to monitor what is going on inside the CADENZA as it processes the audio.

The programming is done by first selecting the condition for when the respective **Probe** LED should be activated, for instance the input level falls below an assigned threshold. Once the condition becomes true the **Probe** LED is lit and held lit for an addition duration of ~0.1 s.

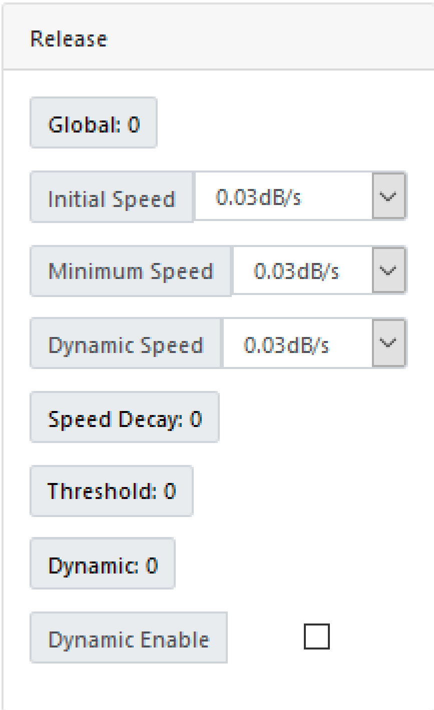
For most events there is an associated *Level X* (dBFS), *Gain Y* (dB) or *Speed Z* (dB/s) that is separately programmed by dedicated controls.

3.9 Attack Controls



Attack is the period over which a compressor will reduce the gain to reach the determined level. Typically expressed in units of time.

3.10 Release Controls



Release is the period over which a compressor is increasing gain to reach the determined level. Typically expressed in units of time.

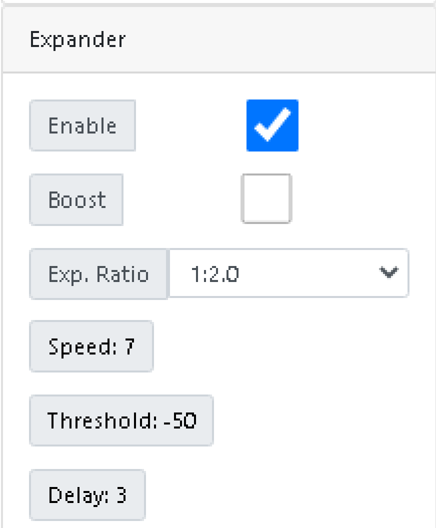
3.10.1 Attack and Release speeds

The attack and release algorithms perform the key functions of the Compressor in the CADENZA.

Each algorithm has some automatic components and some user defined components. There are *Global* speed controls scaling the multitude of settings. When the attack or release is found to be too slow or too fast, enter the appropriate *Global* control and scale the speed up or down.

Please remember that these controls adjust the reaction *speeds*. In terms of time**,** chose a lower percentage to get a longer time and a higher percentage to get a shorter time.

3.11 Expander Controls



3.11.1 Boost

If the Expander fade-down speed is needed to accelerate at very low audio input levels, the *Boost* function should be enabled. For very low *Expansion Ratio* settings this function will lose its effect.

3.11.2 Expansion Ratio

The Expander works like a mirror image of a compressor: The lower the audio input level, the lower the gain.

There is a defined relationship between an input level difference and the resulting Expander output level difference. This is the *Expansion Ratio*. With a 1:2 setting the expander will double an input level difference. A fade-down will be performed. With a low ratio setting the fade-down is gradually performed as the input level slowly drops, preventing a restless gain changing activity at the very weakest sounds. With a very high ratio setting there will be a switch-like expander take-over. Levels below the *Expander Threshold* will all get a very low gain and the fade-down is controlled by the time and speed parameters only. This is in fact a gate function, which is often referred to as a “Silence gate”.

3.11.3 Speed

The Expander fade-down speed is set with the Expansion speed control. The speed is somewhat proportional to the difference between the audio input level and *Expander Threshold* setting. The figures in the control menu are just for the navigation on a **Less/More** scale.

3.11.4 Delay

After entering the Expander mode, the fade-down will come to effect after the delay period set with the Expansion *Delay* control. The delay is inversely proportional to the difference between the audio input level and *Expander Threshold* setting. The larger the difference the shorter delay. The delay is also proportional to the Expansion *Speed* setting. The figures in the control menu are just for the navigation on a **Less/More** scale.

4 Adjusting Parameters

4.1 Gain Stop

Gain Stop parameters are most likely to be changed to prevent excessive gain.

> Every parameter setting is a trade-off. If the *Maximum Gain* is set too low, the desired gain for very quiet sounds may not be achieved. On the other hand, a high setting may allow excessive gain on pieces of audio where it is not wanted. To add a little intelligence to the *Gain Stop* function, these three controls should be used in parallel:
>
> - *Maximum Gain.*
> - *Hold Level.*
> - *Expander Threshold*.

4.2 Adjusting the *Maximum Gain*

If the dynamic gain stops performed by the hold and/or Expander functions are satisfactory, then a new *Maximum Gain* does not need to be set. Should the actual gain be too high at times, adjust the *Maximum Gain* control to the desired absolute maximum possible gain.

4.3 Music process *Gain Stop* parameters

For music with narrow dynamics, the *Talk process gain stops* may be used with lower *Hold level* and *Expander Threshold* settings.

For Classical music, the borderline between the quietest musical sounds and noise is very diffuse or non-existent. A high-level gain stop may be set but may miss the extreme gain when it is really needed. Alternatively, a low-level gain stop may be set, but disturbing noise may be amplified along with the most silent musical sounds.

When setting the gain stops for wide dynamics music, start by adjusting the *Maximum Gain* to +18 dB or more, before using the expander function. For broadcasting, the expander will be very helpful in the joints between music and announcements.

To use the Expander, the *Expander Threshold* should be the first gain stop to set. If it not required, deselect *Enable* in **Expander** menu and adjust the *Hold Level*.

4.4 Adjusting the Expander Threshold

Start by setting the *Hold Level* at –80 dBr (disengaged). The Expander function also incorporates a hold function which is active during the first part of an expander sequence. This is indicated by the **Hold** LED and the **Expansion** LED lighting simultaneously. The Expander's hold function is a little more complex than the normal hold, as it is dependent on both actual input level and actual gain.

Run music material with wide dynamics, containing some very quiet parts, and try to find an *Expander Threshold* setting suitable for the programme format. The *Expander Threshold* will probably land in the level range from –40 dBr to –60 dBr. The actual fade starts when only the **Expansion** LED is lit. The gain meter displays the fade. The ideal is of course that the Expander starts fading as the musicians stop playing. Hold can be active during the quietest parts of the music, but preferably no fade.

> This is a true Expander and not a slow gate, so audio levels just below the threshold will be faded slightly. The deeper below the *Expander Threshold* the more is the gain reduction, according to the *Exp. Ratio* setting.

As the Expander's hold function depends also on the actual gain, the Compressor will have less stops on its way towards large gain values. This may be an advantage when active processing is needed, or it may be a disadvantage when the gain is increasing during short pauses in the music. The latter is for the normal *Hold* function to deal with. Make it active and evaluate the result from this combination.

4.5 Adjusting the *Hold Level*

For music with wide dynamics the *Hold Level* setting could be anywhere within a very wide range depending on the audio quality of the music material and the desired effect. Probably within the –35 dBr to –60 dBr range. The general rule is to adjust it to the lower end of the useful dynamic range of the input audio.

Run wide dynamics music material to find the *Hold Level* setting performing gain stops where needed. Observe the **Hold** LED and listen for noise or ambience being unnecessarily fished up in pauses. Note that a normal hold and an expander hold can be distinguished by means of the LED activating.

4.5.1 Noisy recordings

A high *Hold Level* setting is the way to avoid annoying noise pumping on noisy recordings. Perhaps in combination with a high expander threshold setting. The compressor should be alive only when the music is playing loud.

If the programme format contains both types of dynamics, one must be favoured over the other or a trade-off is necessary. In the latter case, at least leave the *Expander Threshold* at a low setting to avoid an expander fade during quiet parts. It might be better to reduce the compression *Gain Ratio* instead of heightening the *Hold Level*. Work with a combination of these two and the *Global Release* control to slow down the processing. As an alternative approach, the **Less/More** control can affect both ratio and release speed.

> The ideal is to switch between two different presets which are optimised for the two types of music productions. Such a switch will be unnoticeable, even if it is made when the music is playing.

4.5.2 Hold Level for mastering

When compressor gain in a mastering process is not needed, a very high *Hold Level* setting can be a way to avoid audible gain changes.

4.5.3 Adjusting the Maximum Gain

If the dynamic gain stops performed by the *Hold* and/or *Expander* functions is satisfactory, then it is unnecessary to set a new *Maximum Gain*. If the actual gain is too high at times, adjust the *Maximum Gain* control to the desired absolute maximum possible gain.

4.5.4 Adjusting Unity Level and Compression Ratios

Audio compression is performed by amplifying low input levels and attenuating high input levels. Somewhere between these two respective level ranges there is a specific input level that will be neither amplified nor attenuated. In other words, this input level will generate 0 dB gain, which is also called *unity gain*.

In most of the presets, input levels equal to the set Reference Level will have this *Unity Gain*. This very input level is meant to be the average peak level of the audio material.

However, there are reasons for making this *Unity Gain* point differ from the *Reference Level*.

4.5.4.1 Unity Level

When adjusting the *Unity Level* control, move the compressor's fixed point or 0 dB pivot away from the *Ref Level* setting.

The *Unity Level* is set as a relation to the *Ref Level* (dBr):

Unity Level = 0 dBr means: *Unity Level = Ref Level*

Many level thresholds in the CADENZA are set as a relation to the Reference Level. These thresholds are expressed as "**dBr" values**.

0 dBr means a level equal to the Ref Level setting, +5 dBr means a level 5 dB above the Ref Level etc.

The reason is that all functions shall remain the same when the CADENZA is adapted to different input level standards.

4.5.4.1.1 Using the Unity Level

Once the *Ref Level* has been set, there are a few reasons to make the *Unity Level* differ from this setting. In practice the fixed point called unity level is diffuse depending on the actual attack speed. Only a very high attack speed will produce exactly 0 dB compressor gain at a unity level input. But a high attack speed will not produce pleasant audio processing.

The *Unity Level* setting may be used to create a margin for the compressor. By lowering the *Unity Level* setting a few dB, the average peak output levels from the compressor can be maintained at a slow attack speed.

4.5.4.1.2 Using the two compression ratios

With the *Unity Level* and the two ratio controls can be used to obtain some interesting compressor functions. Input levels below the set *Unity Level* will be pushed up according to the *Gain Ratio* setting, while input levels above the set *Unity Level* will be pulled down according to the *Atten. Ratio* setting.

The twin ratio feature of the audio path processor may be used to create some different types of compression. A low *Gain Ratio* may be set for a smooth over-all compression, at the same time as the very loud parts exceeding the *Unity Level* are compressed a lot harder, according to the *Atten. Ratio* setting. In this operation some Limiter parameter most likely may be changed.

4.5.4.1.3 Switch to linear amplifier

A more interesting combination is perhaps the use of a 1:1 *Atten. Ratio* setting. This means that the compressor turns to a 0 dB linear amplifier for input levels above the set *Unity Level*. Only the input levels below the *Unity Level* will be affected by the compressor.

With this ratio combination you may go way down with the *Unity Level*. The fortissimo parts of the music can be free from compression and make the gain start at the required level. The draw-back this time is less over-all loudness.

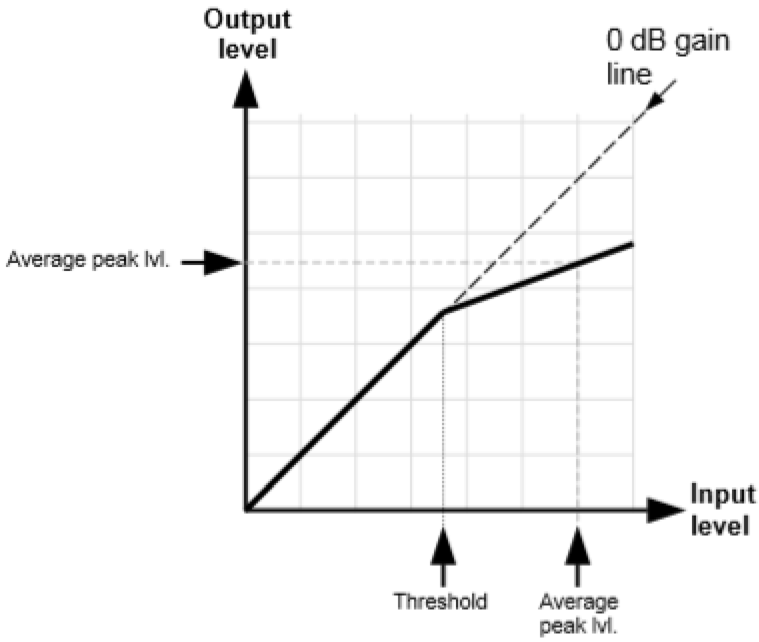
The 1:1 *Attenuation Ratio* principle is used in a special way in many of the presets. Together with a 0 dBr *Unity Level* setting, all peak handling is left to the Limiter.

4.6 The Gain Law

A few basic parameters define the compression gain law. These parameters are found in the **Basic** menu. The diagrams below will clarify the differences between the way of defining the gain law in a conventional compressor compared to CADENZA.

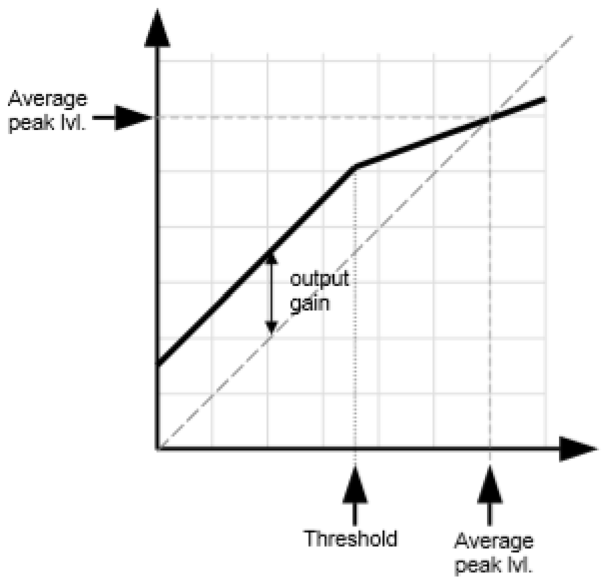
4.6.1 The conventional way

A conventional compressor works with Ratio, Threshold and Output gain (make-up gain). Input levels above the threshold setting are reduced according to the ratio setting, like this:



In this operation the desired reduction of the dynamics is performed, but the total loudness is reduced as well. The average peak level is considerably reduced. Usually, the loudness reduction is not a desired effect, and unexpected from sound compression.

Adding static gain by pushing the fader or turning the output gain. This achieves the desired loud and dense sound, as illustrated:

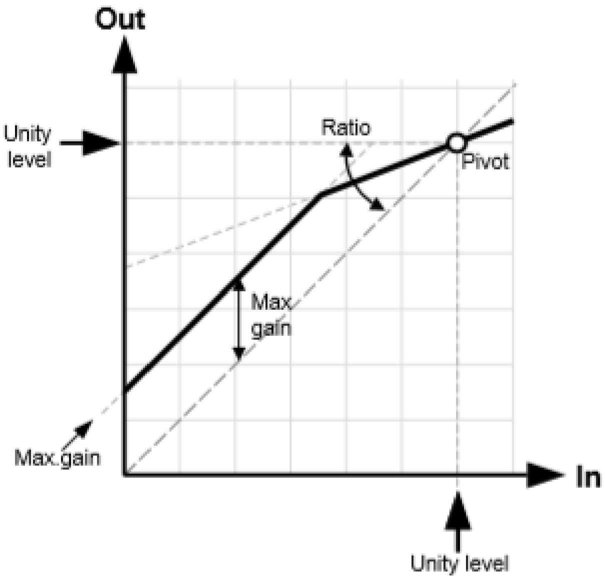


Furthermore, it is likely a constant average peak level is wanting to be maintained when trimming and adjusting the parameter settings. Also, when checking the compression effect by switching it on and off. To accomplish this, repeated output gain or fader corrections are needed.

4.6.2 The CADENZA way

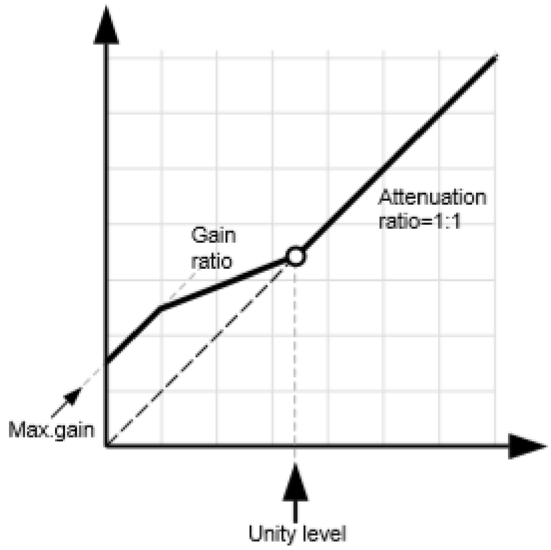
CADENZA works with *Unity Level*, *Ratio* and *Maximum Gain* when defining the gain law.

The *Unity Level* control defines a specific input level that will always have 0 dB gain = unity gain. The compression curve is nailed to the framework of the diagram at this very point. The nailing point works like a pivot. The curve may be reshaped in any possible way by the other parameter settings, but the curve will always go through this point.



Usually, it is wise to match the *Unity Level* to the average peak level of the audio material. Then there will be no unpleasant surprises when tweaking the compression parameter settings. The average peak level will always be under control. Trim the Compression Diagram with the *Ratio* and *Max. Gain* controls. This is much like working with the conventional Ratio/Threshold combination, but with the addition of an automatic output gain corrector. The expected louder sound of the compressed audio is immediately obtained in the CADENZA.

There is a *Threshold* also in this Compressor. This curve knee will slide up and down when the *Ratio* or *Max. Gain* controls are altered. The actual position of the knee is not important as you cannot hear it. The *Ratio* and *Max. Gain* are much more logical parameters to control, when rapidly tuning a compressor by ear.



CADENZA has separate ratio settings below and above the Unity Level pivot. The *Gain Ratio* setting controls the gain below the pivot, while the *Attenuation Ratio* setting controls the attenuation above the pivot. This feature, along with lowering the Unity Level, are the absolute conditions for creating the unique combination of low-level compression and linear dynamics at loud sound passages.

5 SUPPORT

5.1 What to do if you have a problem

Firstly, please ensure that you have followed the installation, connection, and operation instructions in the appropriate User Guide.

Also, check the Troubleshooting section (where appropriate) to eliminate common problems.

5.2 Servicing, Maintenance and Repairs

Please contact your supplier or Factum Radioscape for all questions relating to maintenance and repairs.

Any unauthorised attempt to open, modify or otherwise repair the product will invalidate the Factum Radioscape warranty and may result in the product being left in an irreparable condition.

5.3 Support

For warranty, technical and application support issues, you should initially contact your supplier to check whether your Factum Radioscape product is covered by warranty, extended warranty, or maintenance contract.

At Factum Radioscape, we will make our best efforts to provide prompt and friendly support by phone, and e-mail.

5.4 Support Contact Information

Address:	LABS Triangle, Chalk Farm Road, Camden Town, London NW1 8AB UK
Tel:	+44 (0) 020 7126 8170
E-mail:	support@factumradioscape.com
Website:	www.factumradioscape.com

5.5 Support Requests

When contacting Factum Radioscape for support, please provide as much information as possible about the problem or issue for which you require assistance.

Please provide the following details (where available) in your Fault Report:

- Software Version
- Details of any symptoms or error messages
- Diagnostics information (if available)
- Sequence of events/actions or other circumstances that triggered the problem
- How you are able to identify that there is a problem?
- How you have been able to measure, log or otherwise display the problem?
- Details of the host PC (if appropriate)
- Sample data files (if appropriate)

Less-more	%

BASIC – SPEECH – ATTACK – RELEASE – EXPANDER – LIMITER – DRIVE

BASIC

Unity level	dBr
Gain ratio	:1
Atten. ratio	:1
Maximum gain	dB
Hold level	dBr
Standby gain	dB
Ctrl.smoothing	

ATTACK

Global attack	%
Expert	
Max atk. speed	dB/s
Min att. speed	dB/s
Dynamic atk.	
Atk. algorithm	

EXPANDER

Exp. function	
Exp. threshold	dBr
Expert	
Expansion ratio	1:
Exp. speed	
Exp. delay	
Exp. boost	

LIMITER

Limiter level	dBr
Limiter drive	dB

DRIVE

Output drive	dBr

SPEECH

Control source	
Favourise	
Music level	dBr
Speech level	dBr
SWAP	
Exp. threshold	dBr
Exp. delay	
Exp. stop	dB
Min att. speed	dB/s
Rel.init. speed	dB/s
Hold level	dBr

RELEASE

Global release	%
Expert	
Initial speed	dB/s
Min speed	dB/s
Threshold	dB
Speed decay	%
Dynamic rel.	
Max dyn. speed	dB/s
Dynamic func.	

Preset name:

Design by: *Date:*

Notes:

Less-more	%

BASIC – ATTACK – RELEASE – EXPANDER – LIMITER

BASIC

Unity level	dBr
Gain ratio	:1
Maximum gain	dB
Hold level	dBr
Standby gain	dB
Ctrl.smoothing	

ATTACK

Global attack	%
Expert	
Max atk. speed	dB/s
Min att. speed	dB/s
Dynamic atk.	
Atk. algorithm	

EXPANDER

Exp. function	
Exp. threshold	dBr
Expert	
Expansion ratio	1:
Exp. speed	
Exp. delay	
Exp. boost	

LIMITER

Limiter drive	dB

RELEASE

Global release	%
Expert	
Initial speed	dB/s
Min speed	dB/s
Threshold	dB
Speed decay	%
Dynamic rel.	
Max dyn. speed	dB/s
Dynamic func.	

Preset name:

Design by: *Date:*

Notes:

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