Engineering Bulletin

Level Magic / Leveler / Dynamics - Parameters Description

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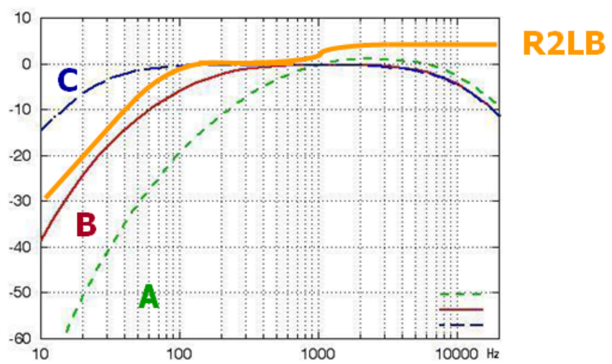
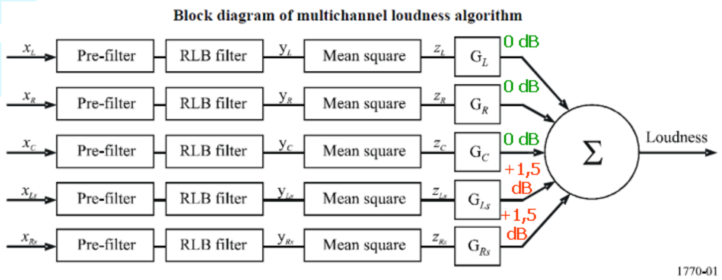
Level Magic

Level — The Junger Audio proprietary, level based process. The aim is to maintain a desired operating level. Curves and algorithms are the **intellectual property** of Junger Audio and will not be disclosed.

ITU-BS.1770-1 (A85/2011) — Loudness based measurement. Several filters and **RMS** weighting are used to get a loudness equivalent result.
Starting from the well known A, B, C weighting curves (DIN-IEC 651), the **ITU** did further research into the relationship of frequencies, their overall levels, their peak levels and the duration of signals to develop the best representation of human loudness perception.
The result was the **RLB** curve [Revised Low frequency B]
The combination of the **RLB-filter** and the Pre-filter is called **R2LB** **[**Secondly Revised Low frequency **B** curve. AKA **K-Weighting** curve :



E.g. **K-Weighting** filters are used as in the example below for the measurement of loudness of a surround signals (LFE must not be included).



ITU-BS.1770-2/-3/-4 — The frequency weighted measurement has been extended by a gating function. The **EBU PLOUD-Group** developed a gating function to exclude quiet sections from the measurement to prevent possible loudness underread. Examples are background noise and atmo which do not add to the loudness perception as much as louder signals do. This gating function consists of an absolute threshold at -70LUFS and a relative threshold 10LU below the absolute-gated level. This involves the necessity to recalculate the whole measurement with every subsequent block. The **ITU** approved the gating function and included it into its recommendations **ITU-R BS. 1770-2** and later.

Important note! Systems working in **ITU 1770-1** mode **do not** feature a **gating** function. Thus its output readings may vary a bit from meters compliant to **EBU R128** or **ITU 1770-2/-3/-4**. Further keep in mind that the gate is only applicable to the integrated or program loudness measurement (from start to stop) and *not used* for short-term or momentary measurements.

EBU R128	This is a **work to rule**, based on ITU-BS.1770. To characterise an audio signal the measures of **Programme Loudness**, **Loudness Range** and **Maximum True Peak Level** are used. The **Program Loudness Level** is normalized to **-23LUFS** [Loudness Units referenced to Full Scale] with a permitted deviation of **+/- 1LU**. The measurement includes a **gating** method as specified in ITU-BS.1770 (summarized in EBU Tech Doc 3341). **Loudness Range LRA** measures the variation of loudness on a macroscopic timescale. It is supplementary to the measure of overall (integrated) loudness. Units are LU. The algorithm for calculating it can be found in EBU Tech Doc 3342. The **Maximum Permitted True Peak Level** of a program during production is **-1dBTP**, measured with a meter compliant with both ITU-BS.1770 and EBU Tech Doc 3341.
ARIB TR-B32	Association of Radio Industries and Business (Japan)
ATSC A/85 (2011 / 13)	Advanced Television Systems Committee (North America, Washington, D.C.)
Free TV OP-59	Free TV Australia Operational Practice
Portaria 354	Ministro de Estado das Comunicações, Brazil

Leveler – **Input Gain** (dB) [+20.0 … 0.0 … -20.0]

The input level can be altered by +/- 20dB to match level diagram needs or for static loudness offset control.

Leveler – Level mode - **Operating Level** (dBFS) [0 … -50]

This is the target level of the whole system. All processes within the **Level Magic** algorithm are designed to aim for the **target level**. It is crucial to understand that the target level is not a threshold and is not a reference for peak levels of any kind. For easier understanding imagine the target level as the balance point or center of gravity of the signal. Level Magic is balancing the signal around this centre, thus achieving a consistent loudness impression for the listener. Single peaks are not affected by this balancing process so that, as far as possible, the natural dynamics of the program are preserved.

Leveler – ITU mode – **Loudness Target** (LKFS) [0 … -50]

ITU has defined the unit of measure to **LKFS** (loudness K-Weighted referenced to digital Full Scale)

Leveler – EBU mode – **Loudness Target** (LUFS) [0 … -50] (Loudness Units referenced to digital Full Scale)

EBU has defined the unit of measure to **LUFS**

Imprtant Note! LKFS and LUFS are different units for the same measure. They are fully compatible.

Leveler – **Time** [10, 20, 40sec. / 1, 2, 5, 10, 20, 40min / 1, 2h]

This controls the speed at which **Level Magic** tries to reach target level. This setting should not be confused with the attack time of a conventional sound processor. As the leveling process is a self-adjusting system this time is not an absolute term but rather an initial value that could exceed the numerical value many times. When setting it, it is necessary to take the overall function of the system into account. Production duties may require faster time settings, while ingest or play-out correction systems may need slower settings.

Level Magic / Leveler / Dynamics - Parameters Description

Leveler – **Max Gain** (dB) [0 … 40]

This parameter controls the **maximum permitted gain change** to reach the target level. It can be useful to limit the maximum amount of gain so as not to overly boost noise and other unwanted signals. The maximum attenuation is not affected by this setting. The system regulates the maximum attenuation adaptively to the signal structure.

Leveler – **Freeze Level** (dBFS) [-20 … -60]

The Freeze Level function holds the amount of gain or attenuation if the signal level drops below this threshold. It works in a similar way to a Hold function in other sound processors. Although this sounds difficult, it is in fact easy to understand with an example. Assuming the process applies a gain change of 10 dB to achieve target loudness, the input level will suddenly drop below freeze level. The gain change remains in its last state until the signal returns above **Freeze Level**. This behavior is different to the Processing Threshold (see below) where the gain change would return to its neutral state if the level falls below threshold. It is necessary to always set Freeze Level above the Processing Threshold to prevent unwanted release behavior.

Transient Processor – **Max Gain** (dB) [0 … 15]

The **Transient Processor** can be **limited to a maximum processing gain range**. Sometimes a hard setting with a very limited gain range can sound more natural than a softer response at full gain range. Adjusting the **Transient Processor** according to the designated overall behavior of the Level Magic process will improve its neutral processing character.

Transient Processor – **Response** [soft, mid, hard]

The **response of the Transient Processor** is a highly **self-adjusting process** which reacts adaptively to the incoming signal structure. Its response can be adjusted in three presets from a more vital to a more relaxed setting but it also depends on the **Limiter Processing** setting.
That means that the overall handling of transients and peaks is determined by the parameters of the Transient Processor **and** the Limiter.

Transient Processor – **Response Boost <boost>**

Response Boost is a function that temporarily increases the Transient Processor's reaction intensity. It raises the compression ratio by a factor of 1.5 for more effective control of signal sections that lie above the target loudness level. For signals below the target level, the nominal Response parameters are applied. Response Boost is intended to be a temporary function that allows users to utilize relatively conservative Level Magic settings for normal operation, and to rapidly increase the processing parameters at the point of transition to a potentially louder than target section (e.g. movie to commercial break). This feature can be triggered either through loading a preset, a parameter event from an external trigger (GPI, Automation) or manually through a button within the Web UI.
It can be triggered once or held on for as long as required. After the trigger is released, the ratio slowly returns back to its nominal value over a period of approximately 10 seconds.

Please note that when triggered by loading a preset, Response Boost is fired once and then released immediately. To hold the function on for a longer period, it must be triggered through a Parameter Event for the **D*APx** platform or a GPI for c8k.

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Limiter – **Max Peak** (dBFS) **/** – **Max True Peak** (dBTP) [-20.0 … 0.0]

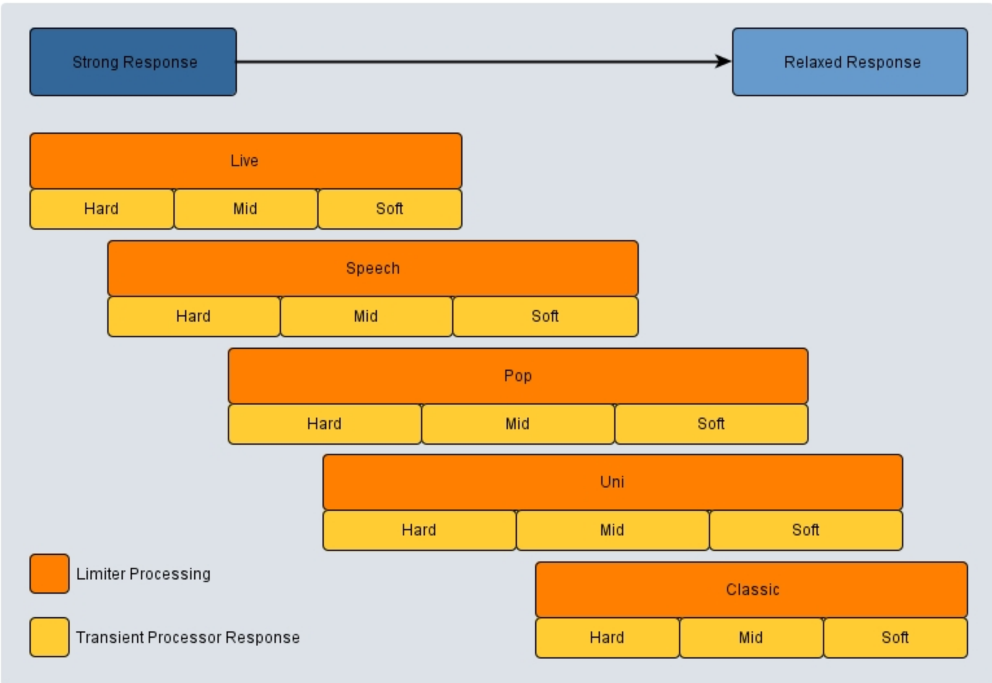
The **Maximum Peak Level** sets the threshold for the system's true peak limiter. Its fast detection system with a 2ms look-ahead time characterizes its response as a full **brick wall limiter**, not only for the obvious sample peaks but also for the hidden inter-sample peaks.
Its precision fulfills all criteria defined in **ITU 1770**.

Processing Profile (Leveler, Limiter) – **Processing** [live, speech, pop, uni, classic]

Although generally speaking our processing will never sound bad but it is possible to further improve its neutrality by selecting one of the five given presets to match the actual content of the processed audio signal:

live	fastest response
speech	fast response
pop	medium response
uni	slow response
classic	slowest response

In older implementations the Processing Profile or simply called Profile is not available separately for the Leveler and the Limiter. In that case selecting a Profile has an effect on the Limiter AND the Transient Processor. The following picture illustrates this interaction between transient processor and Limiter depending on the parameter settings :



Expert

Under normal conditions Level Magic is runs as a 'set up and forget' process with astonishing leveling results and crystal clear sound quality. To improve it further, it is necessary to integrate the system into an automation environment and to adapt some of the settings to the given program material. For systems providing these technical requirements the 'Expert' mode / parameters was introduced.

Level Magic / Leveler / Dynamics - Parameters Description

Expert – **Clear Processing History <clear>**

This is a triggered action that resets the dynamic processing without any release time. Imagine it as a short circuit to the timing circuits of an analog dynamic processor which discharges the whole system and immediately returns the dynamic gain to its neutral state. See picture X for further clarification. This function is useful to reset the process when switching programs (e.g. from movie to commercial breaks). The same function is available to the compressor section (without the 'Initial Dynamic Gain' option).

Expert **– Initial Dynamic Gain** (dB) [-40 … 15]

This parameter directly depends on the 'Clear Processing History' trigger. Instead of resetting the dynamic gain, it can be preloaded to a desired value the moment 'Clear Processing History' is triggered. The preload value is specified by the 'Initial Dynamic Gain' parameter. This helps reduce attack time artifacts, if switching programs incorporates a known and undesired level jump. It is not necessary to exactly predict the level difference between the programs but already helps to set up a few dB in the right direction. It is easy to understand with an example: The level jump from a movie to a commercial break is usually around 6 dB. Resetting Level Magic at the transition point helps to even out the difference. Nevertheless Level Magic needs some attack time to build up damping of 6 dB, which can be audible depending on the program structure. If the process is reset to a damping value between -4 to -6 dB instead, the attack time is much shorter and artifacts fall below the perception threshold. In many cases an Initial Dynamic Gain value of +/-3 dB is sufficient to create transitions with seamless loudness.

Expert **– AGC Recovery** [normal / fast]

All gain changes are processed adaptively to the incoming audio signal. Under normal conditions this adaptive reaction is working fine. However in special configurations it can be necessary to have a faster recovery or release time. Again, an example helps to explain the effect. If Level Magic is configured to work without gain or just a very small amount of positive gain, returning to unity from heavy attenuation can take quite some time. If a very loud part (above target) is followed by a quiet section (right at or below target) the recovery from damping leads to an unnatural 'fade in' effect for the quieter part. To decrease this effect 'AGC Recovery' can be set to 'fast' to accelerate this 'fade in time'. It is accelerated up to fifteen times its normal speed. The result sounds almost the same as if an audio engineer is riding the fader to correct unwanted level jumps and thus is very natural and well accepted by the listener. Please note that this setting is most helpful for setups where no positive gain (AGC amplification) is allowed. The effect works relative to the 'Leveler Time' setting and hence is more obvious for short 'Leveler Time' values.

Expert – Low Level Behavior **– Processing Threshold** (dBFS) [-80 … -70 … -20]

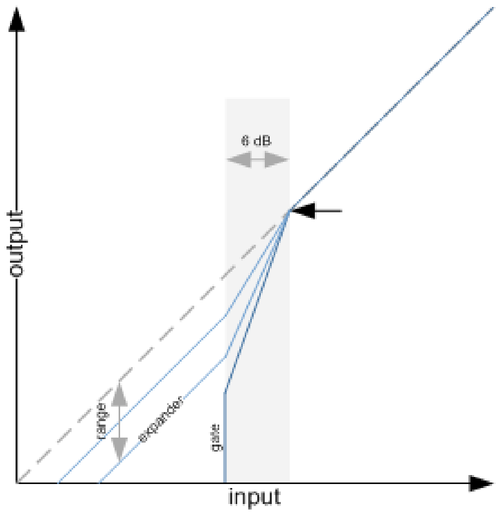
The **Processing Threshold** determines the lowest level to activate **Level Magic** processing. If the signal drops below this threshold all **Level Magic** parameters return to their initial values or remain in their current state, depending on the setting of the Below Threshold Mode. It is recommended to set this threshold slightly above the lower dynamic range limit of the whole audio system you are using or the dynamic range you are intended to broadcast. In many cases it makes sense to just leave it at its lowest value.

Expert – Low Level Behavior **– Below Threshold Mode** [Release, Hold]

In continuous operation the 'Below Threshold Mode' should remain in 'Release'. In this case the dynamic gain slowly returns to its neutral state in case of signal absence. In this mode a returning signal initiates a new processing period with its lead in attack time. However, this can be undesired. In those cases setting the 'Below Threshold Mode' to 'hold' will pause the dynamic processing and freeze the gain at the last valid value until the signal returns. Returning signals are treated just like continuous signals. This function has some similarities to the 'Freeze Level' function but works with a different designation as it is meant to keep processing fluent over periods of signal absence.

Expander

The classic Jünger expander reduces noise and unwanted sounds in a very unobtrusive and effective way. When the signal falls below threshold it is steadily reduced over a level range of 6 dB. In consequence 6 dB below threshold the maximum reduction is reached no matter which ratio is used. In Gate mode, its ratio is set to infinite to one.



Expander – **Threshold** (dBFS) [-60 … -20]

Signals below threshold are reduced, signals above pass unaffected.

Expander – **Range** (dB) [0 … 20]

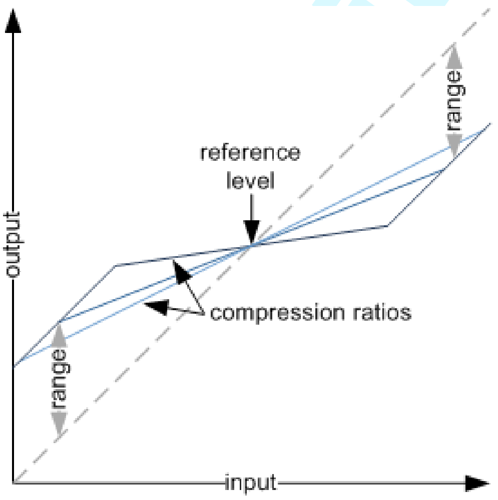
Maximum reduction range. Its value determines the maximum reduction of the input signal. This parameter is sometimes called 'floor', but differs in terminology. A floor level is defined as an absolute value in dBFS, no matter where the threshold is set. Range defines the relative range of reduction in dB below threshold and is thus independent from absolute values. When set to 'Gate', the input signal is muted.

Expander – **Release Mode** [0 … 9]

The release profile controls the timing of the closing of the Gate/Expander. Release mode 0 is very fast even short gaps or signal intermissions lead to gain reduction. On the other end of the scale, 9 is a very slow mode with a relaxed handling of gaps and low level periods. All modes feature the same super fast opening when the signal returns above threshold.

Compressor

All signals below reference level are amplified according to the ratio and range settings, all signals above reference level are reduced in the same way. This is the 'classic' approach of earlier Junger Audio compressor designs.



Compressor – **Reference Level** (dBFS) [-40 … 0]

Not to be confused with threshold, this parameter defines the turning point of the response curve from upward to downward compression (see picture). When set to 0 dBFS, all the signal is amplified according to the ratio and range settings.

Compressor – **Range** (dB) [0 … 20]

This defines the range over which dynamic compression is applied as defined by the ratio setting.. Signals outside of this range are still reduced or amplified but not altered in their dynamic structure.

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Compressor – **Ratio** [1.1 … 4.0]

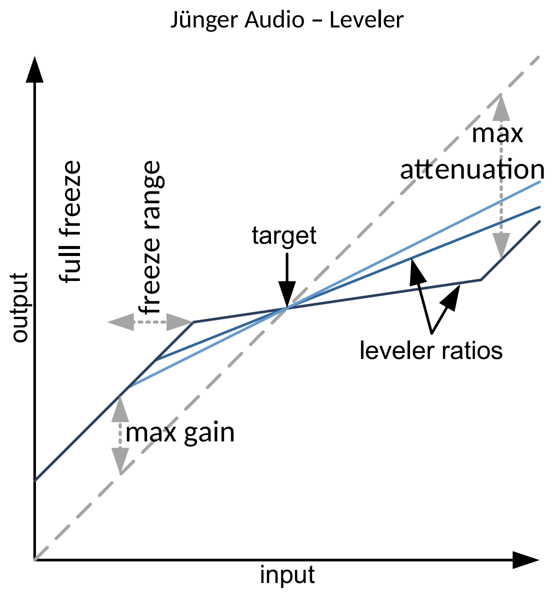
Determines the amount of gain reduction by a selectable ratio. A ratio of 2:1 means that an input level of 4 dB below reference level will result in an output level of 2 dB below reference level. In the same way an input level of 8 dB below reference level results in an output level of 4 dB below reference level and so on.

Compressor – **Processing** [live, speech, pop, uni, classic]

The timing characteristics of the compressor are generated adaptively according to the incoming signal structure. The overall timing can be set up from fast and responsive settings (lower numbers) to relaxed settings (higher numbers) without detailed access to the actual micro timings. The names behind some of the numbers may help to easily find adequate values to your content.

Level Magic / Leveler / Dynamics - Parameters Description

Voice Leveler



Voice Leveler – **Input Trim** (dB) [-80.0 … 0.0 …. 20.0]

Simple input level trim to prevent the Leveler from reducing static offset. Hint: To use the Input Trim without the Leveler, simply reduce its Max Attenuation and Max Gain down to 0.

Voice Leveler – **Max Attenuation** (dB) [0.0 … 10.0 … 40.0]

Defines the maximum attenuation or damping of the Leveler process for signals that exceed the target level.

Voice Leveler – **Target** (dBFS) [-50.0 … -24.0 … 0.0]

This is the balance point of the Leveler. Signals that exceed the target are reduced, signals below target are amplified to bring the overall level to this center of gravity.

Voice Leveler – **Max Gain** (dB) [0.0 … 10.0 … 40.0]

This is the balance point of the Leveler. Signals that exceed the target are reduced, signals below target are amplified to bring the overall level to this center of gravity.

Voice Leveler – **Time** (s/m) [1s … 30s … 2min]

This defines the timing for the Leveler to reach target. Of course this is not an absolute value as it depends on the input level, signal structure, ratio and necessary amount of gain change.

Voice Leveler – **Ratio** [1 … 10 … 40]

As opposed to a classic compressor, the Leveler ratio is normally fixed to infinity to one. With this Leveler you can bring down the ratio to a much lower value to achieve a more relaxed compensation instead of heavy steering towards target. Setting it to 1:1 disables the leveling process.

Voice Leveler – **Freeze Threshold** (dBFS) [-60.0 … -50.0 … 0.0]

Voice Leveler – **Detector Weighting** [Full Range / Proximity / Loudness]

The Leveler features a Side Chain Filter with special characteristics to adapt the leveling process to three major applications.

Voice Leveler – **Detector Weighting [Full Range]**

The Side Chain is not filtered and the Leveler is running on full bandwidth detection.

Voice Leveler – **Detector Weighting [Proximity]**

The Side Chain uses a low shelf filter to compensate for microphone proximity effects.

Voice Leveler – **Detector Weighting [Loudness]**

In modern broadcast production environments the final product is played out in accordance with current loudness standards. Those standards and recommended practices always refer to the output signal and do not consider the condition of the source channels. This is correct, but never the less, it can be very useful to consider loudness for these single channels. With this loudness filtering you can bring the output to a consistent level, based on modern loudness recommendations. The output will integrate seamlessly into your loudness normalized product. In many situations, no additional loudness correction is necessary. This approach is compatible with all international loudness recommendations.

Technical remark: K-Filtering is used as described in ITU-R BS 1770-3.

Voice Leveler – **Low Level Behavior**

Voice Leveler – Low Level Behavior – **Processing Threshold** (dBFS) [-80 … -70 … -20]

The **Processing Threshold** determines the lowest level to activate the **Leveler** processing. If the signal drops below this threshold all **Leveler** parameters return to their initial values or remain in their current state, depending on the setting of the Below Threshold Mode. It is recommended to set this threshold slightly above the lower dynamic range limit of the voice channels. In many cases it makes sense to just leave it at its lowest value.

Voice Leveler – Low Level Behavior – **Below Threshold Mode** [Release / Hold]

In continuous operation the 'Below Threshold Mode' should remain in 'Release'. In this case the dynamic gain slowly returns to its neutral state in case of signal absence. In this mode a returning signal initiates a new processing period with its lead in attack time. However, this can be undesired. In those cases setting the 'Below Threshold Mode' to 'hold' will pause the dynamic processing and freeze the gain at the last valid value until the signal returns. Returning signals are treated just like continuous signals. This function has some similarities to the 'Freeze Level' function but works with a different designation as it is meant to keep processing fluent over periods of signal absence.

Voice Leveler – Low Level Behavior – **Leveler Hold** <Normal> / <Hold>

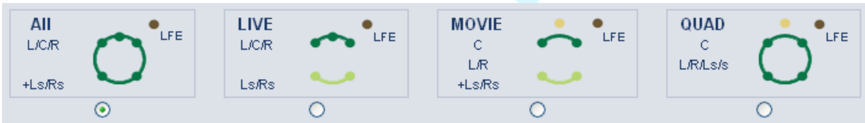
Leveler Hold freezes the levelers dynamic gain and preserve its current state, for as long as the hold function is activated. This can be useful for example to prevent loud sounds like sneezing or coughing from causing leveler action.

Level Magic / Leveler / Dynamics - Parameters Description

General Parameters

General parameters – **Link**

The link function connects all parameters of two channels in a stereo pair. In situations where mono or mid-side transmission is utilized, it is recommended to unlink channels. More importantly it links the control circuits of the processing blocks in order to maintain the sound balance of these channels.
Several **link options** are available depending on the operating and process control modes.
While **4 x 2 mode** is straight forward (linked or unlink), the **5.1 + 2 mode** offers different modes for the surround channels, depending on the device. For **EBU** mode we have two options:
ALL (LFE is not linked) and **ALL+LFE** (LFE is linked).
Below is an example showing the link modes from a **T*AP**, which was set to **EBU** mode, and from a **C8086+,** which was set to **Level** process control.
Dots joined by a line or having the same colour are linked together:



General parameters – **Bit Transparent** [On/Off]

Bit transparency is necessary in cases where non-audio signals are present at the input. A non-audio stream can be any kind of coded transmission, like **Dolby Digital** or **Dolby E**. In Auto mode the process is detecting whether audio or non-audio is present and automatically switches the unit to bit transparency.

General parameters – **Processing Status Monitor** [On/Off]

It is possible to monitor the gain change of the control process. An error status will be provided if the average of the gain change is equal to, or **above**, the **Leveler Range** setting for **more than 10s**.
If this option is turned on, a soft LED for GUI applications will turn from red to green. This status information is combined for all processed channels/programs and is presented as a module (c8k) or device (T*AP, LMx) status to an external monitoring system by sending a **SNMP** trap and/or by firing a **GPO**. The parameter itself is also available for polling.

General parameters – **Loudness measurement**

Junger Audio equipment and software based applications offer several loudness metering displays.
The following items are found on these displays with their abbreviation in square brackets:

Momentary [M]	[LUFS]	sliding time window of 0.4sec. length
Short Term [S]	[LUFS]	sliding time window of 3sec. length, not gated
Integrated [I]	[LUFS]	start/pause/continue/reset measurement, gated, of Momentary values (see EBU-Tech 3341)
Loudness Range [LRA]	[LU]	of the program (integrated measurement)
Maximum True Peak Level Max [TPL]	[dBTP]	during the measurement period