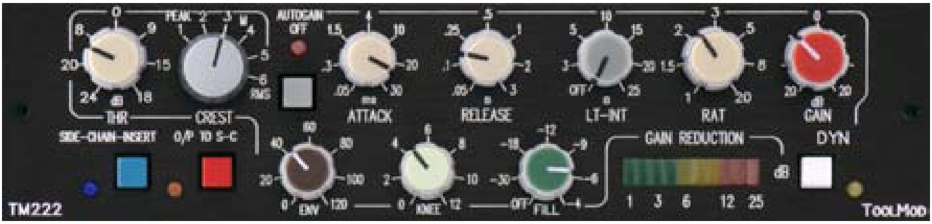
# ToolMod Pro-Audio Module System

## User Manual

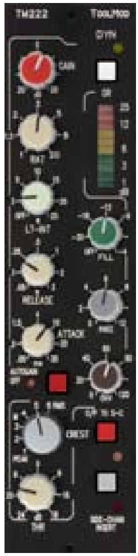
# TM222

## Stereo Mastering Compressor




**The TM222 Stereo Mastering Compressor comes with many additional functions to achieve high loudness gain without the typical 'compressor sound'. The module is best suited for mastering dynamics processing and as master compressor for mixing.**

**The TM222 is available in horizontal and vertical versions. Both versions are electrically identically and differ only in the lettering of the faceplate and the orientation of the control knobs. The module fits into 1U-high and 4U-high ToolMod frames.**



## Content



## Hint

We assume that you are familiar with the basic principles of pro audio devices. Therefore, the explanations in this manual are limited to the special features and functions of the particular module. General information on many pro audio topics can be found on our web site http://www.adt-audio.com

This manual is a supplement to the **ToolMod User Manual** that contains extensive information and safety instructions on the **ToolMod® Pro-audio Module System** by adt-**audio®**. If you don‘t have the **ToolMod User Manual** ask for a copy by email or fax, or download a PDF-version from one of our websites. **It is imperative that you take account of the hints and safety instruction in the ToolMod User Manual that are not repeated in this supplement for a particular module.**

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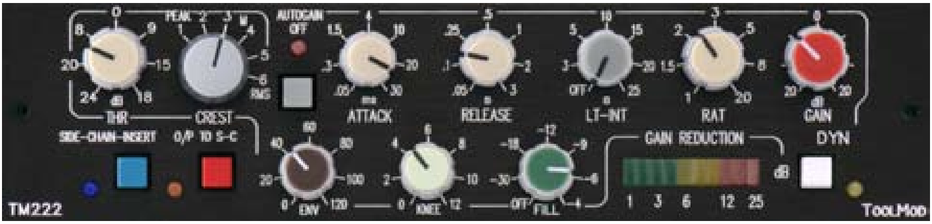
## Principle of Operation

Very flexible feed-forward controlled, fully parametric VCA stereo compressor with calibrated stereo tracking, variable crest factor setting, integrated parallel compression, side-chain insert, and autogain, with optimized regulation characteristics for dynamic treatment with high loudness gain for mastering and mixing.




## Control Elements

**THR - Threshold**
Determines the begin of the compression in the range from + 18 dB to - 24 dB. Below this level no compression takes place. The internally calibrated 0 dB position refers to + 6 dBu ≅ + 4 dBV.

**CREST - determines the behavior of the ac/dc converter**
The 6-position rotary switch CREST determines the basic characteristic of the compression from position 1 / Peak to position 6 / RMS. With CREST set to peak mode, the TM222 works as fast and aggressive compressor that responds to the peak value of the audio signal. Position 6 selects RMS mode, which results in slow and soft compression that is controlled by the loudness of signal. Steps 2 to 5 are interim values between the two extremes. Use Crest to set the basic characteristic of the compressor. In most cases, the interim values 2 to 5 are the best choice. The crest setting also determines the minimum attack and release time. This is only important if you select RMS or position 5.

**ATTACK - determines the attack time of the compressor**
This pot determines the attack time of the compressor in the range from 50 µs to 30 ms. However, the minimum attack time is also determined by the crest setting (see above). The attack time is very important for the characteristic of the compression. Shorter attack times cause more aggressive behavior while longer settings leave the transients unchanged and result in a more natural sounding compression. The range from 1.5 to 20 ms is best suited for mastering and mixing. The attack time refers to a gain reduction of 10 dB.

**RELEASE - determines the release time of the compressor**
The release pot changes the time the compressor takes to fade up after the level drops. With long release time settings, audible ‚pumping' appears. If the release time is too short, low frequency distortion can occur. The release time can be adjusted in the range from 50 ms to 2 sec per 10 dB.

**LT-INT - adds program depending release**

This pot determines if and how much the density of the mix modulates the release time. With low density signals (short peaks only), the LT-INT setting causes no changes while with high density signals (constant high levels) the release time is increased. This function is similar to the ‚Auto-Release' feature of some compressors; however, you can adjust the amount of ‚autogain' instead of switching over to a preset value. All to the left, LT-Int is disabled. The more to the right, the higher is the influence on the release time. Please note that the integration time depends on the setting and can be up to 25 sec. Therefore, it takes some time before the effect of a changed setting is stable. (You can ‚reset' LT-Int just by turning the pot all to the left for a second and than go back to new setting).

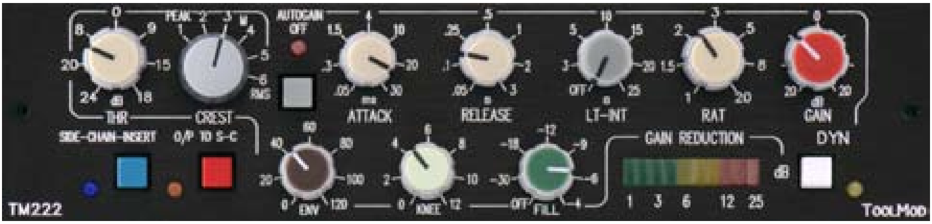
**Ratio**
The Ratio pot determines the relation between input and output level above threshold and therefore the magnitude of the compression. With an input level 10 dB above threshold and a ratio of 2 the output level is 5 dB above threshold. With a ratio setting of 5, 10 dB above threshold is reduced to 2 dB. All to the left, no compression takes place. The maximum ratio is 20 : 1. With this setting, the TM222 works as a limiter. Practical values for mastering are in the range from 1.2 to about 3.

**Gain - controls the output level**
The gain pot only controls the output level of the compressor. It does not change the compression itself. The control range is ± 20 dB. The 0 dB position has a center click and is internally calibrated.

**Autogain - automatic compensation of the gain reduction**
The autogain circuitry keeps the output level constant independent of the THR, Ratio, and Attack setting. With autogain you don't need to readjust the output gain after a setting has been changed. The AUOGAIN-OFF switch disables the function. The reference level for the compensation can be internally adjusted. The factory standard setting is + 6 dBu ~ + 4 dBV. You can specify other levels with the order. Autogain and gain operate independently. You can manually change the gain if autogain is active or not.

**ENV - Envelope, improves low frequency compression performance**
Compressing bass signals with short, pump-free release time settings, can cause distortion due to the regulation within the period time of the bass signal. With short release times, the ripple rejection of the control voltage that actually is the rectified audio signal is low. The ripple on the control voltage causes an additional modulation of the audio signal that results in distortion. This physical effect limits the minimum pump free release time related to the gain reduction and the spectral density of low frequency signals. The envelope circuit improves the performance of the compressor by a special filter that reduces the compression of bass signals and increases the release time with such signal a little.

**Knee - determines the transition into the compression around the threshold level**
The Knee pot determines the ratio in the transition range above the threshold. All to the left at 0 dB, there is no transition range at all. As soon as the level exceeds threshold, it is compressed with the ratio that is determined by the setting of the ratio pot. The knee range can be set to a maximum of 12 dB. With this setting, the ratio increases from 1 : 1 over a level range of 12 dB to the setting of the ratio pot. Using such a setting with high ratio values can make higher compression possible or reduce the negati-




ve side effects. To keep the overall level constant, the threshold is shifted down with 'softer' knee settings. It's in the nature of things that the effect of a soft knee setting is lower if the ratio is small, since the difference between the uncompressed and the uncompressed range is small.

**Fill - integrated parallel Compression**
The integrated parallel compression is controlled by the Fill pot that mixes the uncompressed input signal to the compressed ouput. With higher compression and short, pump-free release time settings, the details of the signal structure and the transients become more and more 'damaged' as a negative side effect of high loudness gain. Adding the uncompressed - and therefore undamaged - input signal to the output can restore the damage considerably; however, adding too much input signal will reduce the loudness gain. The maximum level is - 4 dB; practical values are between approx. -15 and - 8 dB.

**Side-Chain-Insert & O/P to S-C**
The stereo side chain insert point makes it possible to integrate external equalizers and other processing units into the control loop of the compressor to modifiy the behavior of the compressor under regulation. Like the audio inputs and outputs, the side-chain insert inputs and output are electronically balanced and have the impedances, levels, and headroom. The side chain output signal is always present while the side chain insert input is selected by the SIDE-CHAIN-INSERT switch. The O/P TO S-C switches the signal at the side chain insert input directly to the output. Direct control of the settings of external gear is therefore possible.

**Gain Reduction Display**
A LED chain with 10 LEDs and a total range of 25 dB displays the actual gain reduction.

**DYN-Switch**
The DYN switch is the hard bypass for the entire module. If not pressed, the output is switched to the input by relays.

## Base Settings / Quick-Ref

The control ranges of all parameters of the TM222 stereo mastering compressor make many different compressor settings possible - from a fast, peak controlled compression with audible pumping to a soft, loudness oriented compression that maintains the originality. To simulate one of the 'classic' compressors, simply set the controls to the same values and select an appropriate Crest factor. However, the most important use of the TM222 is a compression that achieves high loudness but maintains the originality of the mix, the transients, and also preserves the subjective impression of dynamic.

For this, start with a base setting as follows:

**Base Settings:**

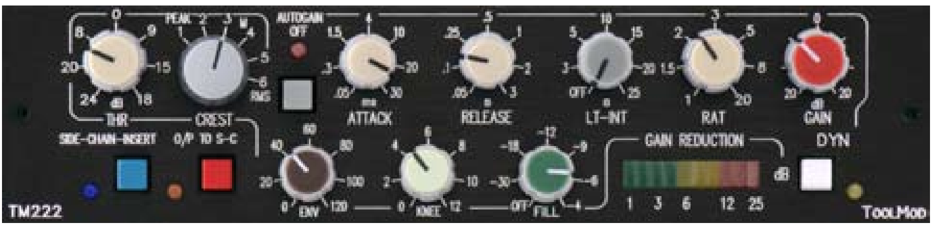
Threshold - ~ - 15 dB ... - 20 dB
Crest - 3
Attack - 6 ms
Release - 0.3 sec
LT-Int - all to the left - **IMPORTANT !**
Ratio - 2
Gain - 0 dB
Envelope - all to the left
Knee - all to the left
Fill - all to the left - **IMPORTANT !**

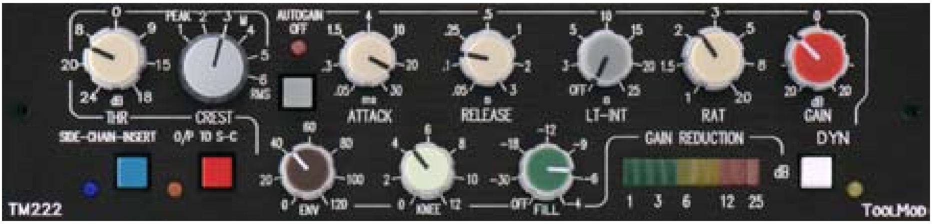
## The Adjustment Process

**1. Threshold and Ratio**
Starting with the base setting above, readjust the Threshold until you get a gain reduction of about 10 dB. The compression should already be a little too high. It is better to use lower threshold levels than higher ratio values. It works best if the compressor is under regulation for the entire mix (which is the case when the actual level never drops below the threshold). If in doubt, turn the threshold pot more to the left and leave the ratio pot as it is or use less ratio to reduce the compression.

**2. adjusting the Attack Time**
Now, turn the Attack pot all to the right. Due to the very long attack time, almost no regulation takes place anymore, since it simply takes to long for the compressor to react. Turn the attack pot slowly to the left until you here the compression come up again and find a setting that lets the transient pass thru without causing regulation but is fast enough to fade up the 'breakdown' of the signal after a peak. With such a setting you avoid severe damage of the transients. Since human ear recognizes 'sound' by the structure of the transients, the compressed mix sounds the more natural the lower the transients are deformed. On the other hand you need to go as close to the point of an acceptable damage to the transients by using a shorter attack time, to achieve considerable loudness gain. In most cases you will end up in the range between 8 and 20 ms for the attack time.

If you switch the compressor on and off after this setting, you should hear a higher loudness without that the character and the naturality of the mix has changed. Make su-






re that you don't have different levels with and without compressor for this test. The autogain should take care about this; however, you may need to fine trim the output level with the gain pot.

**3. optimizing Threshold and Ratio with the new Attack Time**
After adjusting the attack time, optimize the setting of threshold and ratio for best loudness gain and/or to your liking.

**4. optimizing Crest**
Try to get an improvement by changing the Crest setting. Choose the setting that is best suited for the mix you are working on.

**5. adjusting the Release Time**
After optimizing Crest, we need to fine tune the release time. Release time setting is always a compromise between audible pumping (= you can hear the compressor fade up when the level drops) and low frequency distortion, which is a physical effect that appears if the release time is too short.

'Pumping' becomes audible if the release time is longer than the release time of the human ear of approx. 0.3 sec/10 dB with individual and program depending differences. If the release time of the compressor is shorter than this approximate value of 0.3 sec, the audible pumping disappears. Such a setting is absolutely necessary to increase the density - and therefore the subjective impression of higher loudness.

However, with a release time setting in that range another problem appears that limits the minimum release time. With short release time settings and low frequencies, the control voltage, which is actually the rectified audio input signal, is no dc voltage but a dc voltage with overlayed remains of the audio signal. The shorter the release time and the lower the frequency, the lower is the filter effect of the compressor's time constant circuit. The ripple on the dc control voltage causes additional regulation and the additional regulation causes the distortion. This is a physical problem that is not limited to certain compressors.

Set the release pot from the start value of 0.3 sec for a moment to about 1 sec - now you will hear the pumping - and turn slowly to the left to find the point where the pumping just disappears. Ignore any lf distortion for the moment.

**6. compensating LF Distortion with the Envelope Circuit**
The Envelope circuit (ENV) can reduce the low frequency distortion considerably. The circuit combines a special filter that reduces the bass compression with a short and inaudible increased release time for the time a low frequency signal is present. The filter characteristic compensates the increasing distortion toward lower frequencies in

a way that the distortion remains constant over the frequency. The ENV pot sets the start frequency for this filter up to 120 Hz. At 120 Hz the bass compression is reduced to keep the distortion constant at lower frequencies. Envelope is no magic bullet but helps a lot to find a better compromise between pumping and lf distortion. In addition Envelope avoids that the entire mix is modulated by the kick drum and or the bass. So far the principle.

**➔ If you get low frequency distortion or modulation of the compression from kick drum or bass with the 'just no pumping' setting of release pot, just turn the ENV pot to the right until the distortion and/or modulation disappears.**

**7. Trying Soft Knee**

At this point, you can try to achieve a higher compression by slightly reducing the threshold and/or increasing the ratio. Do it and try if you can improve the performance with the Knee pot. The knee circuit changes the transition from the unregulated part of the compression curve to the compressed part, which is 'hard' with the knee pot all to the left and 'soft' with the pot all to the right. With a soft setting, the ratio increases over an increase of the level that is determined by the setting of the pot to the setting of the ratio pot. The maximum transition range is 12 dB.

Please note that you can get a considerable improvement with a soft knee setting only is the ratio is high. With a ratio below or up to 2 : 1, which is the range we are using for this kind of compression, you will get only subtle improvements. The reason is simply that the difference of the linear part of the compression characteristic below the threshold and the 2 : 1 characteristic above threshold is no high enough to cause a big difference in the behavior.

**8. adding parallel Compression**

After trying soft knee, reduce the threshold and/or increase the ratio again and adjust a compression that is a little too high but take care that you make a mental note of the old setting. It depends on the mix if you better reduce the threshold or increase the ratio; just try. At this point the compression is already pretty high and will damage the structure of the signal and the transients - which limits the maximum compression. The remedy is parallel compression. Parallel compression mixes the uncompressed, undamaged input signal to the output. That will cover the damage; however, adding too much of the uncompressed input signal will also reduce the compression and therefore the loudness gain.

Use the Fill pot to add the input signal. Start from all to the left and increase fill until you get a better result but check by switching on and off if you still have an appropriate loudness. In most cases the best setting is somewhere between - 20 and - 8 dB. The maximum level is - 4 dB.

**9. LT-Int - adding Program depending Release Time Modulation**

You should now get a strong compression with considerable loudness gain that still sounds natural and that appears to have dynamic due to the long attack time, which leaves the transients as they are. You can now use the LT-INT pot to modulate the release time by the density of the signal. The LT-Int (long time integration) circuit is a second release stage that has a fixed attack time of 0.3 sec (like a VU meter), while you set the release time with the pot. The circuit increases the static release time that is set by the release pot if the density of the signal is high and leaves it unchanged with a low density. Use LT Int carefully to loosen the tight compression. If you add too much LT Int you prevent the compressor from filling the 'breakdowns' in the signal and you loose loudness gain. With an appropriate setting you can usually reduce the static release time (with the release pot) without causing additional low frequency distortion.

ATTENTION - due to the long integration time of up to 25 sec, there is a delay before the circuit settles to a new setting. Change the setting in small step and wait some seconds before you check for the result. You can reset the circuit when you turn the LT Int pot all the left.

**10. Checking the Compression**
It is very important that you check the settings by switching on and off permanently and that you always check if the compressed signal and the uncompressed signal have the same levels. Even a level difference of 1 or 2 dB changes the impression of loudness a lot.

**11. Autogain**
Although Autogain is a very useful feature that keeps the output level constant when you change the compressor settings, there are some limitations.

An analoge computational circuit calculates the gain reduction from the settings of the threshold, the ratio, and the attack pots. While autogain works at a very high precision of the less than 0.5 dB over the full range with fast attack time setting, longer attack time setting may cause higher deviations. With longer attack times, the output level depends on the level of the short peaks that pass the compressor without causing regulation. Therefore, the resulting real gain reduction differs from the calculated value that results from the setting of threshold and ratio. Without an additional dynamic circuit, it is not possible to calculate the precise gain reduction since it depends on the randomness of the mix combined with the attack time setting. Such a circuit would change the entire characteristic, which is not useful.

The autogain circuit in the TM222 compensates the attack time setting by empirically determined correction values. Depending on the peculiarities of a particular mix the error might be up to 3 dB over the entire range of the attack pot. However, with the small differences that are made during the optimization of the setting, the differences are negligible. If you notice a significant level difference after the initial setting of the attack pot, just use the gain pot for compensation.

Autogain does not compensate an increase of the level caused by the Fill pot and the slight differences that can be caused by the different settings of the Crest switch.

## Special Features

Some additional features are possible

**Exchanging Equalizer and Compressor in ToolMod Mastering-Sets**
If the Stereo-Mastering-Compressor TM222 is used in combination with a stereo equalizer TM204, TM205, or the 4-band shelving eq with adjustable steepness TM208, two additional patch cables and the side chain insert of the compressor can be used to re-

verse the order of equalizer and compressor without actually repatching the cables. The only requirement to use this feature is to place the compressor post equalizer. Arrange eq and compressor in a way that the outputs of the equalizer are connected to the inputs of the compressor.

The inputs of the equalizers TM204, TM205, and TM208 are connected to the TRS jacks OUT-b in parallel. If the equalizer INPUTS are connected to the side chain insert inputs of the compressor, the regulation takes place controlled by the input signal of the equalizer as soon as the side-chain-insert switch is pressed. In this case the setting of the eq does not affect the regulation of the compressor which results in exactly the same effect that you get if you place the eq post the compressor. The only little downside is that the autogain circuit cannot compensate the level offset that is caused by the setting of equalizer. This offset, that is determined by the difference between the input and the output level of the eq can be easily compensated with the manual gain pot.

The two additional cables are short TRS to TRS cables.

1. connect TRS OUT-b, EQ compartment, left channel with
   TRS IN-b, Compressor compartment, left channel

2. connect TRS OUT-b, EQ compartment, right channel with
   TRS IN-b, Compressor compartment, right channel

With these connections, the switch SIDE-CHAIN-INSERT selects the order of equalizer and compressor:
SIDE-CHAIN-INSERT not pressed - EQ -> Compressor
SIDE-CHAIN-INSERT pressed - Compressor -> EQ

Some notes:
It is not necessary that eq and compressor are installed next to each other or in the same frame. If the audio cabling connects the eq outputs with the compressor inputs, this feature works.

**Feedback Regulation**
The TM222 is a feed forward controlled VCA Compressor. It is possible to operate the compressor with feedback regulation instead, which changes the characteristic of the regulation. To switch over to feedback operation it is necessary to connect the outputs of the compressor to the side chain insert inputs. With these connections, the side chain insert switch selects feedback regulation instead of feed forward regulation.

**Cabling for Feedback Regulation**
Since the outputs of the compressor are wired to xlr male connectors only, two special split cables are necessary. You can order these cables from us or make it yourself. You need two short Y cables with an xlr male on one end and an xlr female and a trs plug on the other two ends.

The following connections are necessary:
1. Output left (OUTa) to side chain insert in left (INb)
2. Output right (Outb) to side chain insert in right (INb)

The compressor outputs are available on the xlr female plugs of the Y cables.

After pressing the SIDE-CHAIN-INSERT switch the compressor operates with feedback regulation.

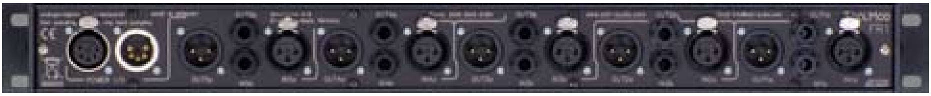
**Important Notes for Feedback Operation:**
The control range of the Ratio pot is different with feedback regulation. The maximum Ratio with the Ratio pot all to the right is 3 : 1.

With feedback regulation, the O/P TO S-C switch causes an internal feedback from output to input.

**DO NOT USE THE O/P TO S-C SWITCH IF YOU USE FEEDBACK REGULATION!**

# Connectors

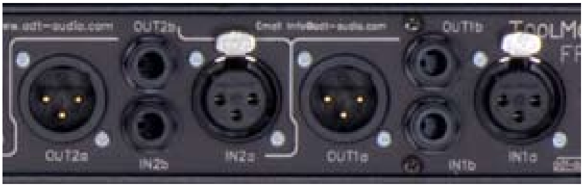
The TM222 is a 4U ToolMod module that uses two adjacent module compartments in a 1U-high frame or two module compartments above each other in a 4U-high frame. The inputs and outputs use the xlr connector of the two compartments. The TRS jacks are used for the side chain insert inputs and outputs.



## Connector Positions in the 1U-high and 4U-high Frame

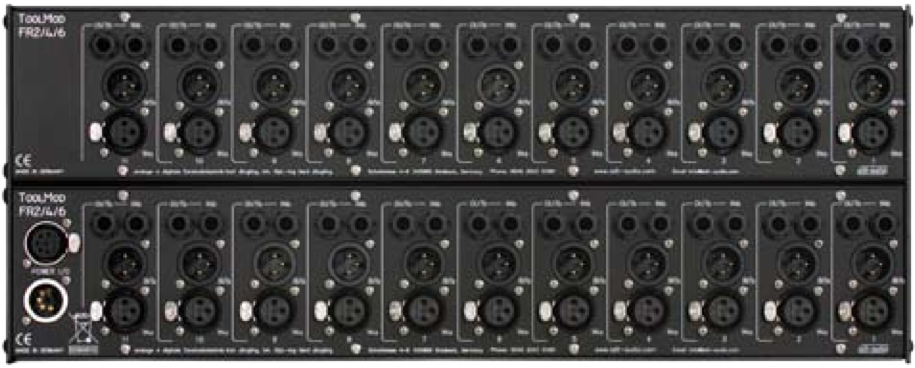
The above image shows the rear panel of the of a 1U-high frame. If the module is installed in the compartments 1 and 2, the connectors are allocated as follows:

Input left: XLR IN1a
Output left: XLR OUT1a
Input right: XLR IN2a
Output right: XLR OUT2a




Side-Chain Insert Input left: TRS IN1b
Side-Chain Insert Output left: TRS OUT1b
Side-Chain Insert Input right: TRS IN2b
Side-Chain Insert Output right: TRS OUT2b

In the 4U-high frame, the location of the connectors is turned counter clockwise. The right channel connectors are located on top; the left channel connectors below.



## Technische Specifications

| | | |
|---|---|---|
| **Format** | ToolMod Module Size 4U | |
| **Versions** | TM222h - horizontal faceplate<br>TM222v - vertical faceplate | |
| **Power Supply** | Tool-Series Standard<br>Supply Voltages +/- 25 V and + 48 V Phantom (not used in this module)<br>Current Consumption +/- 160 mA *) | |
| **Inputs** | balanced, grounded (electronically balanced), Stereo | |
| | nominal Level | + 6 dBu - Gain |
| | maximum Level | ≥ + 30 dBu - Gain |
| | Input Impedance | 20 Hz - 20 kHz, > 10 kΩ |
| | nominal Source Impedance | ≤ 50 Ω |
| | CMRR | 15 kHz > 65 dB, typical 75 dB<br>1 kHz > 80 dB, 40 Hz > 90 dB |
| **Outputs** | balanced, grounded (electronically balanced), Stereo | |
| | nominal Level | + 6 dBu |
| | maximum Level | ≥ + 30 dBu |
| | Source Impedance | 20 Hz - 20 kHz, < 50 Ω |
| | Load Resistance | ≥ 1200 Ω for $P_{max}$ +30 dBu,<br>≥ 600 Ω for $P_{max}$ + 27.5 dBu,<br>≥ 300 Ω for $P_{max}$ + 22 dBu |
| | Load Capacity | ≤ 6 nF \|\| 2 kΩ @ 20 kHz THD = 1 %, + 30 dBu<br>≤ 15 nF \|\| 2 kΩ @ 20 kHz THD = 1 %, + 26 dBu<br>≤ 20 nF \|\| 2 kΩ @ 20 kHz THD = 1 %, + 22 dBu |
| | CMRR (IEC) | > 40 dB, 40 Hz - 15 kHz |
| **Gain** | internally calibriert to 0 dB +/- 0.3 dB without regulation,<br>Fill all to the left, Gain @ 0 dB<br>Gain control range ≥ +/- 20 dB<br>(the Gain pot controls the output level and does not alter the compression) | |
| **Frequency Response** | 3 dB limits < 10 Hz to > 150 kHz (without regulation)<br>Power Bandwidth for Headroom ≥ + 30 dBu from 10 Hz to > 50 kHz<br>Linearity ≤ ± 0.2 dB without regulation from 20 Hz to 50 kHz | |
| **Phase Respone** | 20 Hz-20 kHz < +6/-10° | |
| **THD** | ≤ + 28 dBu, 40 Hz ... 20 kHz, < 0.1 %,<br>max. THD @ + 30 dBu < 1 %<br>(without Regulation - THD under regulation is determined by the frequency, and by the attack- and release-time settings and the actual gain reduction) | |
| **Crosstalk** | ≥ 70 dB, 40 Hz ... 15 kHz | |
| **unweighted Noise** | <= -91 dBu without regulation @ 0 dB Gain<br>(RMS, 22Hz-22kHz, Ref: 0 dBu = 775 mV) | |
| **weighted Noise** | <= -95 dBA without regelation @ 0 dB Gain<br>(AVG, DIN-A-Filter, Ref: 0 dBA = 775 mV @ 1 kHz) | |
| **Dynamic Range** | >= 121 dB without regulation @ 0 dB Gain,<br>(relating to RMS S/N value 22Hz-22kHz) | |
| **Control Voltage Rejection** | ≥ 70 dB | |

| | |
|---|---|
| **Threshold** | Range + 18 dB bis - 24 dB, referred to + 6 dBu = 1.55 V ~ + 4 dBV<br>0 dB position internally calibrated |
| **Ratio** | Range from 1 : 1 (no compression) to<br>1 : 20 (Limiter Operation) |
| **Crest** | changes the Characteristic of the ac/dc Converter<br>6 Position Stepper Switch<br>1 / Peak - Peak Conversion with very short Integration Time<br>6 / RMS - RMS Conversion with long Integration Time<br>2 bis 5 / M - interim Values between Peak and RAMS |
| **Attack** | Attack Time Adjustment for 10 dB Gain Reduction<br>Range 0.05 ms to 30 ms<br>(with very short attack time settings, the attack time is determined by the Crest setting in pos. 5 & RMS) |
| **Release** | Release Time Adjustment for 10 dB Gain Reduction<br>Range 50 ms to 3 s |
| **Envelope** | Function to improve the LF distortion with short release time settings.<br>The Pot determines the upper limit frequency from 0 = Off to 120 Hz |
| **Fill** | parallel Compression, adds the uncompressed input signal up to a level of - 4 dB |
| **Knee** | influences the Transition Range into regulation around the Threshold Level<br>0 dB, ‚hard Knee‘, 12 dB, ‚soft Knee‘,<br>the Ratio is approximated over a maximum range of 12 dB to the setting of the Ratio Pot. |
| **LT-Int** | long Time Integration with 0.3 sec Attack Time and adjustable Recovery Time up to 25 sec.<br>Increases the (static) Release Time depending on the density of the Signal. |
| **Autogain** | automatic Gain Correction depending on the Threshold, Ratio, and Attack Settings<br>active at Threshold Setting below 0 dB = + 6 dBu (factory Standard)<br>AUTOGAIN OFF disables Autogain |
| **Side Chain Insert** | electronically balanced Insert Output and Input with the same Data as the Audio Outputs and Inputs. Ermöglicht die Einbindung externer<br>Allows inserting external gear to modifiy the characteristic of the compression, activated by the SIDE-CHAIN-INSERT switch |
| **Output to S-C** | Control Function for the Side Chain Insert<br>Switches the Side-Chain Insert Input directly to the output of the compressor to allow to check the setting of the external gear. |
| **Display** | Gain Reduction Display with 10 LED‘s, Range 1 dB to 25 dB |
| **Bypass** | Hard Bypass by Relay |

*) The maximum current consumption is the current consumption under real-world operating conditions; i.e. outputs loaded with > 5 kΩ at standard a/d or d/a converter levels of ~ + 18 dBu. If the outputs are loaded with the minium load resistor of 1200 Ω and an output level of + 30 dBu the current consumption increases by 30 mA per output. The standard current consumption should be used to calculated the necessary capacity of the power supply unit.

## Postface and Disclaimer

This manual contains general information on the adt-audio® module system ToolMod®.

By no means does this information represent guaranteed particular characteristics or results of use. The information in this manual has been carefully compiled and verified. Due to our policy of continuous product improvement, we reserve the right to make product changes without prior notice. All specifications are subject to change without notice.



## Trade Marks

adt-audio® and ToolMod® are registered trademarks of analoge + digitale Tonstudiotechnik Karl Juengling. All other trademarks are the property of their respective owners.

## CE Declaration of Conformity

Manufacturer: Fa. Karl Juengling
Type of Equipment: Audio Signal Processor
Product: ToolMod Pro-Audio Module System,
consisting of:
Modules, Mounting Frames, Power Supply Units and Accessories
Compliance Engineer: Gerd Juengling
Test Basis:
EN50081-1:1992, EN50082-1:1992, EN61000-3-3:1995, EN60065:1993 Class1, EN61000-3-2:2000, EN60065:2002, EN55013:2001, EN55020:2002, 73/23 EWG; 93/68 EWG

We hereby declare that the construction of the ToolMod system complies with the standards and regulations listed above.

## Environmental Protection

This product can be recycled. Products bearing this symbol must not be thrown away with normal household waste. At the end of the product‘s life, take it to a collection point designated for recycling of electrical and electronical devices. Find out more about return and collection points through your local authorities.The European Waste Electrical and Electronic Equipment (WEEE) Directive was implemented to dramatically reduce the amount of waste going to landfills, thereby reducing the environmental impact on the planet and on human health. Please act responsibly by recycling used products. If this product is still useable, consider giving it away or selling it.

**WEEE-Registration: DE 59049716**

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