Level Rodeo

A STEP-BY-STEP APPROACH TO FIXING LEVEL PROBLEMS EARLY IN THE MIXING PROCESS USING *REACOMP*

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1 Preface

Disclaimer: prolonged exposure of your signal to compressors and other plugins might have harmful acoustic side effects. In case of doubt consult the manual, reference book or mixing engineer.

Any time I sit down in front of my DAW for a mix I run into the same hurdle – getting the raw initial levels right. It turns out I have forgotten all the great tips and tricks I have read or seen, particularly when it comes to compression. All gone.
I decided to wade though all that again, highlighted pieces here and there, made written notes that hold all the necessary information in one place, which culminated in this compendium.

This appeared to be so useful that it would be a pity not to share it.
Starting from simple word pad notes, I soon decided to make a written, printable document instead of opening a thread in the forum, since readability is of paramount importance. Web browsers have very limited structuring and formatting capabilities, so longish documents become quickly unmanageable and indeed, unreadable. With a well thought about style sheet the document became a joy to read, which is particularly useful for technical documents.

Far from being an expert on the subject. I rather call myself a novice in the lights of those people who have acquired years and years of experience and who are the source of my information. I do however have a technical background and can understand the technical concepts well, and – most important here – I can put a document together.

This article contains as little theory as possible, it is rather written as a cookbook approach to identify and deal with typical level problems one meets early in the mixing process. The goal is to get your volume levels right for a rough mix. It is also not about perfection. Striving for perfection before one starts means getting nowhere. There are already too many big shots that are trying to humble us with their perfectionism. However, there are plenty of links given in the back for those who wish to go on.

This text is dedicated to those people at Mixcraft Forum, who are always willing to help. May it help others too. It is my hope at least.

If anybody wants to collaborate or detects any errors, or anything is perceived missing, or if something is deemed in conflict with law by any means, please contact me under aquataur9999(at)gmail.com and I will do the best to rectify this.

Helmut Gragger, The Tyrol, Austria, 2016-2022

2 Foreword to ed.2

During the course of deeper research on the parameters of compressor icons like LA-2A, 1176 and the like, I stumbled over a video by Dan Worrall [Worrall2019] who postulated that **Reacomp** was severely flawed in several respects, to the point of rendering the intended models unusable.

This was a severe set-back and I considered giving up upon the subject, since no other (affordable) compressor was in sight that would qualify as *versatile compressor engine* by offering all those parameters freely variable.

I evaluated contemporary contenders such as **Fabfilter's** and **Toneboosters's** compressors, but they failed to qualify as the task specific tool I was looking for, because they introduced yet another level of magic and abstraction and thus – loss of control. This is by no means meant to devalue them, because I can see that they can be very powerful tools in the right situation.

Fortunately, the issue with **Reacomp** could be resolved and its reputation restored, at least as the feed-back peculiarity goes.

The first part of the book (ed. 1) is practically unchanged (with minor additions), but several chapters have been added in ed. 2 that get practical by putting **Reacomp** to good use.

3 Preparing Tasks

A few housekeeping tasks have to be performed before ever thinking about compressors or even other tools, which is why they demand to be mentioned briefly. A compressor is not a magic wand you slam on that fixes all your loudness problems. Don´t be sloppy on that or a good result will be in peril. Following the guidelines Mike Senior gives for housekeeping in his book is a good idea. It gives us a rough idea of *if, where* and *when* you need a compressor and prepares the grounds:

3.1 Timing and tuning

Now this is not strictly compressor related, but let’s mention it for the sake.
Get your instrument´s timing and tuning sorted out before starting. [Senior2012, chpt.6]. Fix it *now* or it will never happen.

3.2 Identify the important parts in your song

> *„If you don´t have a goal, reaching it will be impossible. If your vision of a goal is vague, so will be the result. Define milestones, it is easy to get lost on the way, distractions are manifold.”* Helmut Gragger

Be as precise as possible in this stage and don´t be sloppy here. Verse 1 may have different dynamics than Verse 2, same for choruses (= macro dynamic). Set markers to identify those regions. [Senior2012, chpt. 8.1.2]

Use *multing* if necessary [Senior2012, chpt. 5.3.2]. Different sections of the same instrument may need different treatment. This is easy if your tracks are sliced up into sections, delimited by markers.

3.3 Begin with the most important part

...and work yourself downwards. Make every succeeding part a little less dramatic so that you end up with a song that increases in tension towards the apex. [Senior2012, chpt.8.1.1]
Michael Stavrou follows a different approach:

> *„Beginning with the bass and drums will quickly provide a (...) foundation of rhythm and power."* -Michael Stavrou [Stavrou2003, p.175]

3.4 HP filtering (removing low end garbage)

Remove low end garbage before it enters the compressor. This not only steals space from your mix, but (worse) does excite the compressor. [Senior2012, chpt. 8.2.1]. Note that we are *not* talking about side-chain filtering.

3.5 Setting coarse volume levels

> *„Setting proper volume levels is much more important than EQ or auto-dramatization (...)."* -Justin Niebank [Massey2009]

Getting into this would be beyond the scope of this article, but this is probably the **most important** section of this writing. A few approaches are given in [Senior2012, 8.2].

3.6 Listen to what your faders tell you!

If throughout a song a fader´s level seems unstable, meaning you are forever chasing the right setting (hence the title - Rodeo), and cannot find a satisfying position, it is probably time for a compressor.

If your fader is unstable, like you want to rise the volume of a guitar because you think it is too low, but then it is too loud again, and if this is *constant throughout the track*, then the answer may not be a compressor at all. The problem may not be excessive *micro dynamic* or *macro dynamic* movement, but *frequency spectrum.*

This subject not been directly addressed neither by Senior, nor by Izhaki, but Cochrane speaks about it in a video : [Cochrane2016-2]

> *„An equalizer is a* smart volume knob, it only acts upon a specific frequency range." - Graham Cochrane [Cochrane2016-2]

It might just need a *specific frequency range* boosted rather than the track volume.

Chances are, at this early stage those relatively simple measures will suffice.
Take your time, take a break, change speakers, lend an ear or listen to reference material and then revisit your levels. Bypass the effect and see if the situation has improved. Listen to it *in context* with the other tracks. Whatever you choose as a cure for the unstable fader, make absolute sure this is needed, since the best treatment is none.

> „Taste you meal before adding salt" - Darryl Swann [Massey2009]

The subsequent guidelines are a step-by-step approach to breaking down level problems into repeatable scenarios. This is a proven approach, and it might be the only one we have.

It is very important to *try to identify the problem first,* then look the bull into the eyes to catch his horns.

And yes....

> *„Some people put compressor on stuff only because they are told that's what professionals do. Even acclaimed professionals admit regularly that they still don't get exactly how compression really works."*
>
> *„Rule #1: Forget about multi band compressors until you are a genius at single band compression…And even then, keep forgetting them."*
> - Christopher A. Dion [QuantumMusic2015]

3.7 Manual gain riding

> *„With vocals, the engineers on classic recordings also routinely rode the fader in real time while tracking, either into or out of the tracking compressor — a highly skilled operation now largely unheard of among st younger engineers, but which can subsequently enable smoother sounding compression at the mix stage."*
> - Mike Senior [SOS-Senior2009]

Not quite the same thing, but similar: nothing speaks against some gentle manipulation of your track's *volume automation* lines on problematic parts. This makes it much easier for a subsequent compressor.

3.8 Bus compression

Talking about this subject at this early point in the mixing process may seem inappropriate, but omitting to make the right decision early enough can have grave consequences later on.

Compression on the master track is a heavily disputed subject.
As a matter of fact, the mastering engineer (which may well be coincident with you who read this) *will* put some sort of compression on the whole mix.

Keep in mind that compressors in series not just add compression onto each other, their effect is *multiplicative,* meaning if you for example have a 4:1 compressor on a certain part and then another 2:1 compressor on the bus, the total compression would be 8. This can easily lead to over compression and a choked mix altogether. We will attend to that lateron.

So if you plan to use bus compression, you better turn it *on* at this point.

4 Essential Compressor Concepts

At this early stage of mixing, we are not after perfection, particularly with compression. We want to set the controls of a compressor with authority, but we aught to know what they all do first. For the sake of versatility and economy, the universal **Reacomp** is used, which is full-featured and thus has the potential to embrace the spirit of many character compressors.

4.1 Ratio and its relationship to threshold

> *„Logic has it that the higher the ratio, the more the compression applied and the more obvious its effect.* ***One important thing to understand is that the degree of compression diminishes as we increase the ratio.****"* - Roey Izhaki [Izhaki2012, p. 294] (accentuation by the author)
>
> *„Lowering the threshold means more compression, while lowering the ratio has the opposite effect. Often these two controls are fine-tuned simultaneously, while lowering one is followed by the lowering of the other. Ditto for rising."* - Roey Izhaki [Izhaki2012, p. 295]
>
> *„Put another way, lower threshold means* ***more is affected,*** *while higher ratios means* ***more effect on the affected****"* - Roey Izhaki [Izhaki2012, p. 296] (accentuation by the author)

For containing levels (see later), which has a high threshold, this means we may choose a high ratio, say, 10:1, for balancing levels we choose a moderate ratio of, say, 3:1 and for more aggressive compression (like loudening) a low ratio of 1.4:1 might be suitable. [Izhaki2012, p. 295] About the virtues of low levels also see [Reardon2013]

4.2 Attack and release

> *„The transient is where the magic happens…"* - Bootsy (Variety of Sound)

While this may be true, at this stage in the mix, as mentioned earlier, we do not want to get lost in endless fine-tuning of attack and release time settings.

> *„On an SSL console, the compressor has a switch for the attack time. In one position the attack time is fast and in the other position the attack time is slower. Essentially the switch assumes that an engineer only needs two settings: either an attack time that will compress a transient or an attack time that will allow the transient to pass through. In general, this can be a helpful way to think about using a compressor's attack control. Even though a variable knob can be used to set a wide range of attack times, it can be much simpler to ask the question, "Transient? Yes/no?" when choosing your attack setting."* - Eric Tarr [Tarr2014]

On an SSL Type G console this translates to 3ms vs. 30ms [ProAudioFiles2016].

> *„(...) Whether it's 10ms, 100ms, or 1,000,000ms - we don't care!*
> *Instead I want you to think of Attack and Release in terms of being either FAST or SLOW. (...)*
> *From now on, I want you to pretend there are only two settings on your Attack and Release knobs:* ***Almost the Fastest, and Almost the Slowest.***
> *(...)*
> *And if you imagine that both knobs only have two settings:* ***Almost Fast (AF) and Almost Slow (AS)*** *then you really only have 4 combinations of settings* ***(AF Attack + AF Release, AF Attack + AS Release, AS Attack + AF Release, AS Attack + AS Release).****"*
> - Graham Cochrane [Cochrane2016-1], (accentuation by the author)

For the moment, let´s adopt Graham´s Law. This gives us four combinations to choose from. **Reacomp**´s default settings are thus AF Attack, AF Release. We can save the other combinations in presets so we can quickly cycle through them to hear their impact.

4.3 Peak vs. RMS

Since the compressor used may have a choice, we need to know about this setting.
Even if it does not provide a choice, it is good to know what behavior it has to maybe rule one candidate out for a specific task.

> *„In a way, the RMS averaging is a form of attack and release that is applied before the threshold stage. Having a strong link to the way we perceive loudness, RMS is very useful for instruments with a less percussive nature, such as vocals. (...)*
>
> *Peak can work better on percussive performance."*
> - Roey Izhaki [Izhaki2012, p. 306]
>
> *„However, in most of these devices, the effective threshold also changes*[1] *when you change the RMS window or the crest factor adjustment."*
> - Bob Katz, from a private e-mail conversation on setting RMS windows.

For hard and fast numbers, not much could be found at the time of writing. **Reacomp**´s default value is 5ms, which would pretty much be PEAK behavior[2].

For an opto-type compressor, much bigger window sizes, such as 50ms or more can be expected[3]. For our exercise, let us adopt **5ms** (or less) **for PEAK and 50ms** (or more) **for RMS**. User experience welcome.

1 If using ReaComp, you will notice a reduction of the input level displayed when you increase the RMS window size. This meter therefore shows the input level a bit deeper in the side-chain and not the untreated signal itself, which is an option and not a defect.
2 A common peak meter has an integration time of 10ms
3 A CU reading meter, which is commonly referred to as loudness meter, has a 300ms integration time.

4.4 Side chain control

> *„Low frequencies tend to trigger compression more than high frequencies (...).”*
>
> *„A high pass filter on the side-chain can bring about more uniform compression of broadband material.”* - Roey Izhaki [Izhaki2012, p. 307]

Reacomp default setting: full range.

> *„Golden Trick #1 : Use side chain filters on single band compressors in order to treat the dynamic specifically to the frequency range where the problem is.”* - Christopher A. Dion [QuantumMusic2015]

4.5 Knee

> *„The soft knee principle (...) enables smoother transitions between no treatment and treatment (...).”*
>
> *„Soft knee for more-transparent compression. Hard-knee for more effect.”* - Roey Izhaki [Izhaki2012, p. 284f]

Definite values are unknown at the time of writing, but we can resume the 10% and 90% strategy adopted earlier. This would give 0 for a hard knee, 4dB for a soft knee and 20dB for a *very* soft knee. Definite values that character compressors exhibit are unknown.

> „*The lower the knee span, the more evident the compression, but very high knee settings will limit our ability to push the levels high. We generally try to have a knee span a wide as the action range of the input signal;"* - Roey Izhaki [Izhaki2012, p. 311]
>
> „*One issue with soft-knee is that the compression starts earlier (below the threshold) and so does the attack. Since the attack starts earlier, less of the natural attack is retained. Therefore, soft-knee might not be appropriate when we try to retain the natural attack of sounds, and with it longer attack times might be needed." - Roey Izhaki [Izhaki2012, p. 286]*

4.6 Other parameters[4]

Reacomp offers more parameters, like

- **Precomp:** their name for *look-ahead,* such as for limiter applications, leave default (0)
- **Classic Attack:** refers to a previous version for backwards compatibility. Ignore. Setting: default (off)
- **Auto release:** default (off). Although a similar function can be found on some classic units, its behavior may be different.
- **Auto Make-Up**: default (off). In auto mode the plug-in calculates an average value with an unknown algorithm that may or may not be a valid representative for the necessary make-up gain. Some engineers discourage from using this function.
- **Detector Input:** side-chain relevant, leave default (main input)

We will need to harness those parameters if we want to encompass the behavior of some well-known compressors. In chapter 6 I have made an attempt to do so.

4 On Mixcraft forum, I originated a thread that tries to capture the essence of character compressors with *Reacomp*. Chapter 6 dwells on this.

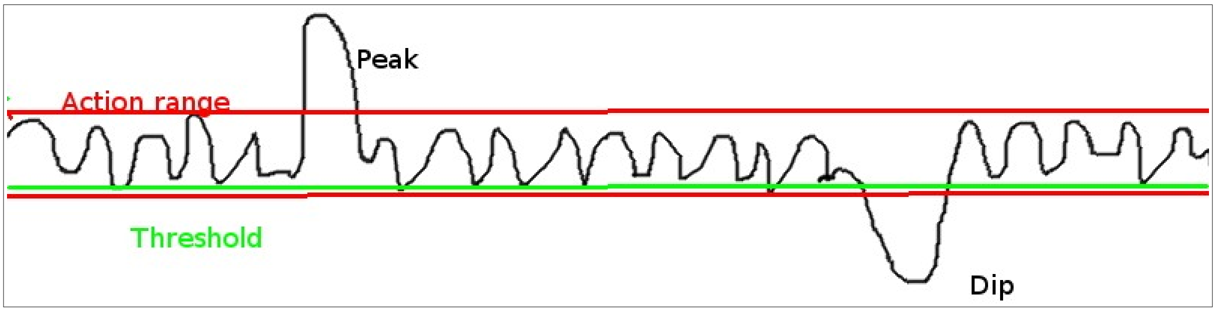
5 Determining Compression Tasks

Once we have landed at the insight, that we do need some sort of compression or transient shaping, we can pinpoint the cause of the problem and choose an appropriate tool.

Novices like the author and even advanced users sometimes have a hard time finding initial settings for a compressor, since they do not know what to listen for. Fortunately, for basic level manipulations, there are three common compression scenarios that may be used as a good starting point: [Izhaki2012, chpt. 16, p.292f].

5.1 Containing levels, i.e. containing peaks

Symptom: an otherwise balanced signal has a few single[5], but extreme excursions (transients) up and down from a somewhat constant action range that poke out (micro dynamic).



Note: The following scenarios approach compressors on a generic, waveform-based mode. Izhaki calls this the “action range”. (Mis-) use your evaluation copy of TB´s **compressor 4** or FF’s **pro-C2** (all timing controls to fastest, ratio=1) to yield a reading similar to the sketch above, which allows you to make decisions about setting initial values. This is an excellent starting point.

5 Note that we are not speaking about shaping recurring peaks like from a snare. This falls under transient shaping. See later on.

> *„Containing peaks is concerned with preventing levels from exceeding a predefined limit – a job for a limiter really. We might want to contain the louder down beats of a strumming guitar , although these might not peak."* - Roey Izhaki [Izhaki2012, p. 293]

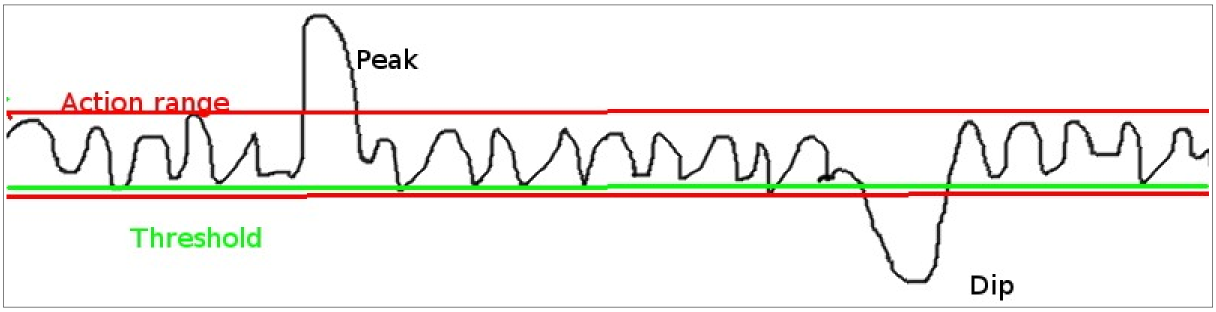
We thus look at a setting of a limiter[6].

> *„To make a compressor a true limiter, we must have peak-sensing, hard-knee and zero attack time."* - [Izhaki2012, p. 331]

Common knowledge would set

- **Threshold:** set at the bottom of the peak, probably where the top of the action resides. Earlier we have determined this to be a high threshold setting, so we can use a use a high ratio.
- **Ratio:** try 20, no need to be extreme
- **Attack and Release:** AF, no need to be extreme
- **Knee:** pretty hard (1dB).
- **RMS window:** peak (try 0 to 5ms).
- **Side-chain HP filtering:** none. We are talking about *micro dynamics* and don´t care about full range signal effects.
- **Make-up Gain:** none. We are not changing overall loudness.
- **Precomp (look-ahead):** Makes sense. Try 20ms

5.2 Balancing levels



Symptom: the level fluctuations of the signal are problematic. A stable fader position cannot be found over the piece of music (macro dynamic)„The threshold is set to a moderate level

6 We are not competing with a limiter here, just giving boundaries to a peak with authority. There is no reason to be as extreme with the settings as a limiter is.

right above the dip, but below all other portions of the signal. The idea is to pull everything above the dip down toward it (...)" - Roey Izhaki[Izhaki2012, p. 293]

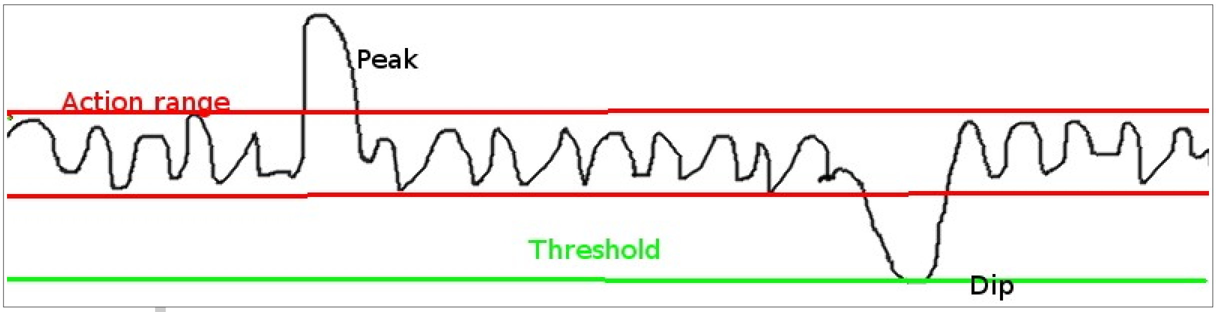
> *„Balancing of this sort usually happens very early in the mixing process, so as to make the tracks more manageable and instill some sense into their relative levels. Usually we are not very fussy at this state about perfecting the compression – rough settings would suffice to give reasonable balancing. As we progress from coarse to fine, we give more attention to the individual compression applied on each instrument."*
> - Roey Izhaki [Izhaki2012, p. 308]

Procedure similar to [Izhaki2012, p. 310]

- **Threshold:** set at the bottom of the action range. Earlier we have determined this to be a moderate setting, so we use a moderate ratio
- **Ratio:** start with 3.0
- **Attack and Release:** AF for less percussive instruments (such as vocals), AS for percussive instruments to keep dynamics.
- **Knee:** soft (4dB). We want to be transparent.
- **RMS window:** dependent on the instrument. Likely RMS (50ms) for vocals and peak (5ms) for percussive instruments.
- **Side-chain HP filtering:** recommended. Play with settings for a more accurate result.
- **Make-up Gain:** as needed to make the resulting loudness perceptually equal.

5.3 Loudening

Symptom: an instrument needs more overall loudness to create a more powerful impact.



> *„The low threshold is set at the base of the signal, so all but the very quiet levels get compressed."* - Roey Izhaki [Izhaki2012, p. 293]

Procedure similar to [Izhaki2012, p. 311f]

- **Knee:** very soft. We want to be transparent. Start with 20dB or the action span.
- **Ratio:** max[7]. Although this is against our earlier postulate (of wanting a low ratio due to the low threshold to avoid over compression), gradual limiting will happen due to the soft knee. Only the highest peaks will encounter a noteworthy ratio.
- **Threshold:** set at the base of the whole signal.
- **Make-up Gain:** identical to threshold, but reverse in magnitude.
- **Attack and Release:** AF for less percussive instruments (such as vocals), AS for percussive instruments to keep dynamics.
- **RMS window:** RMS (try 300ms), we want to conceal the compression effect.
- **Side-chain HP filtering:** recommended. Play with settings for a more accurate result and reduced pumping.

5.4 Reshaping transients and refinement of rough early settings

A special case of track fader instability appears, when an instrument (like a snare hit) seems to poke out in the mix and pushes the instrument too far to the front. If cases like that cannot be resolved with static fader corrections, a compressor can help.

> *„Compression can be used to bring instruments forward and backward in the mix.”* - Roey Izhaki [Izhaki2012, p. 314]

Not only compression, but also expansion and gating can be used for this task, so this is considered *advanced* by the author not only technically. Since stuff like that comes later in the mixing process, it does not belong here.

However, since setting of attack and release times is inevitably entangled with reshaping of transients and thus modifying tonality, we may have to revise our previously casually chosen settings. The author suggests the following procedure:

- adhere to our AF/AS policy (bottom 10% of the control range and 90%) [Cochrane2016-1].
- For a given scenario (e.g. balancing levels), save the setting as preset.
- Save the same scenario with four combinations for attack and release as Graham Cochrane describes it.
- Cycle through those combinations quick and take the one that fits best.

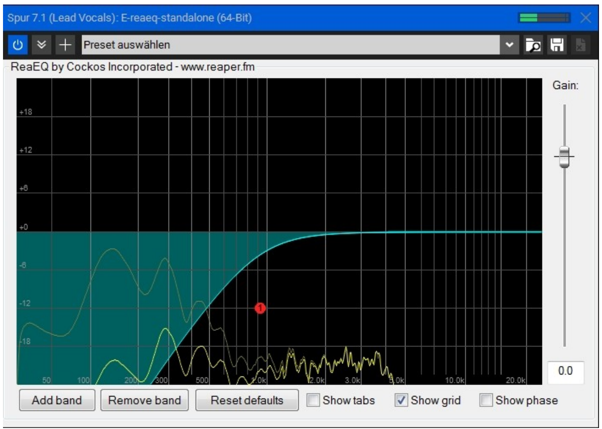
7 Compressors with ratios beyond 10:1 behave much like a limiter. A signal overshooting 40dB is reduced to a resulting 4dB above threshold. [Izhaki2012, p. 294]

5.5 Setting the side-chain HP filter

No approach would be complete without mentioning the HP filter (sometimes termed: *low cut*) in the side chain. (This is not to be confused with a front-end HP filter that removes low-end garbage as mentioned in chpt. 3.4)

In audio, the most prominent components in the spectrum are low frequencies, they excite the compressor first and foremost. Consequently, this parameter needs to be set properly to guarantee a compression that is balanced over the signal's frequency range.

A quick method is is shown in [Sknoteaudio] The method uses a spectrum analyzer display to identify those frequencies and cut them to size accordingly by setting the side-chain's HP filter. A genius approach. Many modern EQ's have a spectrum display, but if you installed the **Reaplug suite, Reaeq** will serve perfectly.



Loop over the section of interest and turn on a first order HP filter. Vintage units have nothing but that in the SC. The brown curve is the full-range signal. As you can see, the bass frequencies are dominant. The compressor will react to them but knock *all* frequencies down.
The yellow curve represents the filtered signal. Move the corner frequency up until you feel the frequencies are evenly balanced.
Take the cutoff frequency you gained and use this a first approach for the side chain.
(Don´t forget to remove the filter again.)

5.6 Joining compressors (serial compression[8])

What if one compressor is not enough? We may find that a single compressor is not capable of fulfilling the tasks required. For example, we might want to balance levels to stabilize our fader, but those nasty peaks are getting in the way. If we manage to tame them, the rest is over compressed. The answer is simple: stacking two (or more) compressors (series), instead of one single instance of a compressor, whose presence is all too obvious.

Although this might sound an advanced subject, it is not[9]. We can simply join two compressors in series. Throughout the course of this writing it would appear a logical sequence in stacking compressors to limit peaks first, and then sort out balance and loudness.

8 Even a single compressor setup can be called serial compression, since the compressor is in series with the signal.
9 Accordingly Izhaki only spends half a page on it.

Caveat:

> *„Most engineers do not realize that Ratios are multiplicative, not additive. (...) If you compress a voice during recording 10:1 end then in the mix again at 4:1 you don´t get a 14:1 but 40:1."* - Mike Stavrou [Stavrou2003]

Izhaki apparently favors the limiting type compressor first (containing levels / peaks), and balancing levels later. This is also the method that seems logical to me, since the second compressor can do its job better without being excited by excessive peaks.

5.7 Joining compressors revisited

As usual, many roads lead to Rome. Bard Reardon (not that the author has ever heard of him, but he as a good web site on the subject[10]) for instance tackles the task the other way round: loudening, balancing levels, containing peaks.

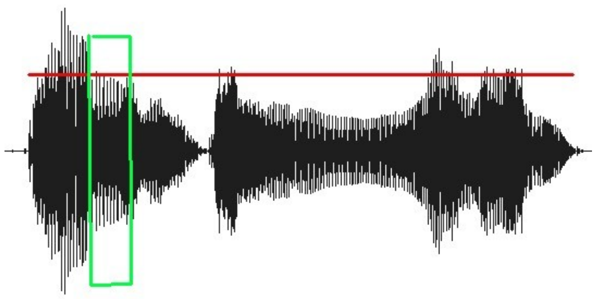
> *„As the signal passes through the chain, the first compressors work more on the lower levels and the last compressors work on the peaks and clamp down more on the attacks.*
> ***First compressor**: Low ratio (2.00:1), moderate attack and release. Let some of the initial attack in the signal sneak through. We're aiming to get some compressor action working in the lower dynamic range.*
> ***Second compressor:** Higher ratio than compressor #1 (4.00:1), slightly faster attack and release. Clamp down just a little more on the signal attacks and let the higher ratio work on the mid-level ranges of the dynamics.*
> ***Third compressor:** Higher ratio (10.00:1, or even 15:00:1). In this one we're setting up the compressor to behave more like a limiter. This one has pretty fast attack and release times and is mostly just catching the rogue peaks in the track."* - Bart Reardon [Reardon2013]

Notice the similarities in the settings!
Reardon describes a method there how he scans for the peaks that is pretty ambiguous because it is dependent on the DAW. I will describe my own method that should work on all DAWs.

10 At the time of writing for ed. 2 this website is no longer accessible. I don´t know of the remains of Bart Reardon.

5.8 Finding the Peaks

For this method, I use the sequence outlined in the Izhaki method in chpt. 5.6.



Look at your sound in a waveform display (sound editor). You can identify some base line (red line) and what you consider rare peaks from the baseline upwards. We want to practically limit the latter. Unfortunately we normally don´t see the absolute level, but this is not necessary.

Loop over the small section to the right of the peak that has the level you want to achieve after compression. Adjust the first compressor so that it just does *not* register (note: set SC filter first). Verify that it does compress with this and other similar peaks, adjust until satisfied. If too much compression results, take down the ratio.
Proceed with the subsequent compressor (RMS type) as described earlier. I find I don´t need a third one. If a medium threshold produces too heavy a gain reduction, I lower the ratio.

5.9 Character compressors

So far we have dealt with improving dynamics to make them more palatable inside a mix in a rather practical way. But what about *sound*, what about legendary character compressors that impart their sonic signature onto the signal?

Truth is, for the purposes used, we are not concerned about *sound* too much, although it does not hurt to use them. Incidentally, typical leveling compressors like the LA-2A are RMS (opto) type and that fast (peak sensing) ones like the 1176 (FET) are used for shaping or clamping transients, while being not restricted to that. Unfortunately through technical heritage (and purpose?) some of their (implicit) parameters are unknown and you are always at the mercy of some preset.

5.10 Right brain side approach to compression

Mike Stavrou is known to have sometimes rather unconventional approaches to things.
He looks at compression from a different aspect, more from the feeling, or rather, from ear.
This procedure[11] can be used to fine-tune attack and release times, if the above procedure does not yield satisfactory results. But there is no reason why both methods cannot coexist peacefully.

11 [Stavrou2003] Chapter 8, Cracking Compressors, p.119ff

Attack fine tuning

- **Release** = fast
- **Ratio** = max
- **Threshold** = sensitive (presumed: somewhere in the action, so that the needle of gain reduction does return to zero frequently and there is not permanent static compression)

Loop over a region of interest. Bring the threshold down and fiddle with the attack knob. Listen to its the front edge if the sound. Ignore the pumping of the fast release. The fast release will make the effect of the attack more obvious.

Release fine tuning

- **Attack** = as found
- **Ratio** = max
- **Threshold** = unchanged

Loop over a region of interest. Bring the threshold down and fiddle with the release knob. Listen to its the tail the sound. Play with release until you feel the groove of the music is supported. Ignore the metronome settings, trust your ears.

Ratio fine tuning

- **Attack** = as found
- **Ratio** = as found
- **Threshold** = unchanged

> *„(...) take the ratio and lower it as far as possible, without loosing the effects you created with your Attack and Release settings.”*
> - Mike Stavrou [Stavrou2003, p. 124]
>
> *„Well, as you raise the Ratio, the sound will become firmer (and smaller) and as you lower the Ratio, it becomes softer (but bigger).* ”
> - Mike Stavrou [Stavrou2003, p. 125]

Remember that low rations mean a much bigger compression.

> *„One important thing to understand is that the degree of compression diminishes as we increase the ratio."* - Roey Izhaki [Izhaki2012, p. 294]

Threshold fine tuning

With all those settings, adjust threshold so, that the of gain reduction meter does return to zero frequently and there is not permanent static compression, and you still achieve the desired amount of compression.

The novice asks: What is the *desired amount of compression*? We see, there this method becomes hairy again.

5.11 Gain reduction

Until now it has not yet be addressed, what amount of gain reduction we are striving for. Yeah, Mike you were letting slip of the truth.
What amount of gain reduction do we want, light (0-4dB), moderate (0-8dB) or heavy (0-30dB)? Don´t forget, consecutive compression processes accumulate *by multiplication.* A gain reduction of 3dB that is later compressed again another 3dB (no unusual scenario) will amount in a nearly 10dB compression[12]!

The idea is (and now we are back to start...) to *apply as much gain reduction as needed to make our fader setting becomes stable*.

> *[insert a compressor] „...and then ask yourself, if the level of the track within the mix is less unstable."*
>
> *„But even if you think, this is the case, you should try to move the threshold control up again to check, if you can live with less compression[13]."*

12 Due to the logarithmic behavior of our ear -3dB is just audible, -10dB perceived as half as loud.
13 This is not the original text, but a loose translation from German back to English.

> *„Even if you managed to achieve adequate level relations, you may find, that signal processing did less than beautiful things to the tone of your instrument (...).”*- Mike Senior [Senior2012, p. 183]

If your gain reduction meter seems lazy or stays in the red for prolonged periods, static compression (equivalent to gain plugin) takes place [Izhaki2012, p. 290].

> *„In order for compression to occur, the amount of gain reduction must vary over time and the gain reduction meter must move, and the faster I moves the more compression takes place.”*
> - Roey Izhaki [Izhaki2012, p. 290]

Definite numbers are not available, but Graham Cochrane for example always is conservative with levels. So start with a 3dB reduction.

Gain reduction is a product of all our parameter´s settings. Since for the above mentioned leveling tasks we may not wish to change the threshold (and thus the range of the affected material) or the timing values, we can only change *ratio.*

5.12 Parallel compression

No writing about compressors would ever be complete without mentioning *parallel compression,* a.k.a. *NY compression.* Although this is may appear an advanced technique, it uses the same principles as explained earlier and it might be applied early too, if you need to louden a signal and find that through series compression you invite too many artifacts (and you always do).

> *„The idea of parallel compression is quite simple – instead of bringing high levels down, we bring the quiet levels up.”*
>
> *„Parallel compression lets us make sound even bigger while retaining their dynamics.”*- Roey Izhaki [Izhaki2012, p. 319]

All compressors discussed up til now were *downwards compressors.* Parallel compression is *upwards compression* or *uplift compression* [SOS2013]; the compressor is *in parallel* to the signal. Its output *adds* to the untreated signal. The output of a parallel compressor on its own would probably defy all rules of transparent serial compression, but that is not its purpose.

Katz describes two common scenarios[14]:

> *„I have found two ways to tackle parallel compression. The first is the transparent approach (...). Here the compressor acts as unobtrusive as possible. It does not produce obvious audible artifacts and hardly any or no loss of transients."* - Bob Katz [Katz2010, p.166]

Stealth parallel-compression

- **Threshold:** set at the base of the action, so all but the very quiet levels get compressed
- **Ratio:** 2.5. As mentioned initially we can afford very low ratios with that low a threshold
- **Attack:** as fast as possible. This compresses the transients of the parallel signal and thus leaves the transients of the original signal intact
- **Release:** medium values. Best work 250-350ms
- **Knee:** no mention, but likely hard for the reasons given before and the small ratio.
- **RMS window:** peak-sensing, works best for transparent settings
- **Precomp:** to aid a fast attack, some look-ahead is in order. Try 20ms
- **Side-chain HP filtering:** no mention.
- **Dry:** this slider is normally on -inf for serial configurations. Move it up to -3dB[15] to let the unaltered input through. Leave on zero if using a send channel.
- **Make-up Gain ("Wet"):** in this case it is not make-up gain. This control controls the amount of compressed signal that is added to the untreated signal. Subtle compression is achieved with -15 to -5dB. Don´t overdo it. If you use a send channel, use its volume slider for mixing in the desired amount of fattening.

This method could be used as an elegant substitute to one or more of the series compression methods described earlier. Note the similarity to serial method "containing signals".

Parallel-compression for tonal shaping and emphasis

Now this an advanced sound shaping technique, which is outside of the scope of this paper, but fits in here beautifully.

14 Translated back to English from German
15 Remember you are adding volume to the unaltered input.

> *„The second approach (...) is a method to achieve emphasis or punch without harming the loud peaks, or to warm up or highlight the music´s low to mid levels. The parallel-emphasis-compressor raises the medium levels effectively, where the meat of the music lies (...). This parallel technique can fatten the sound sometimes better than a normal compressor, because it concentrates on the medium levels, without affecting the highest peaks*"
> - Bob Katz [Katz2010, p.167]

- **Threshold:** set at the center of the action. The resulting gain reduction is 5-7dB, sometimes only 1-3dB
- **Ratio:** set to taste. Adjust together with wet level for the aggressiveness desired. High Ratio levels together with a hard knee can introduce artifacts.
- **Attack:** begin with medium value (125ms). Too fast an attack will suppress transients
- **Release:** adjust together with attack to taste. Make sure the settings support the groove of the music and the pressure you want to achieve.
- **Knee:** no mention. In other articles I have seen a knee used with higher ratios.
- **RMS window:** RMS. Try 100ms or more[16]. Katz states that the best sounding fattening parallel compressors are RMS driven.
- **Precomp:** not necessary.
- **Side-chain HP filtering:** no mention.
- **Dry:** this slider is normally on -inf for serial configurations. Move it up to -3dB[17] to let the unaltered input through. Leave on zero if using a send channel.
- **Make-up Gain ("Wet")**: Due to the smaller gain reduction (compared to the stealth method) higher levels of compressed signal can be added to the original signal (up to -6dB). Again, don´t overdo it.

Again, using **Reacomp** is much recommended for this application. It has a "wet" control (normally make-up gain) and a "dry" control – the original signal. This lets you mix both of them together freely without the threat of comb-filtering[18].
Note however, that creating a separate *send-track* containing this and other manipulations is a much more comfortable and powerful solution. In this case the "dry" signal is zero.

This approach to parallel compression just barely touches upon its surface, but you can go to town on the subject. Since everybody rants about parallel-compression today (and probably right so), it will be easy to find ongoing information.

16 As mentioned earlier, definite numbers from commercial units are not known at the time of writing.
17 Remember you are adding volume to the unaltered input.
18 [Izhaki2012, p. 163] comb filtering frequently appears when combining related, but time-delayed signals.

5.13 Ducking with a Compressor

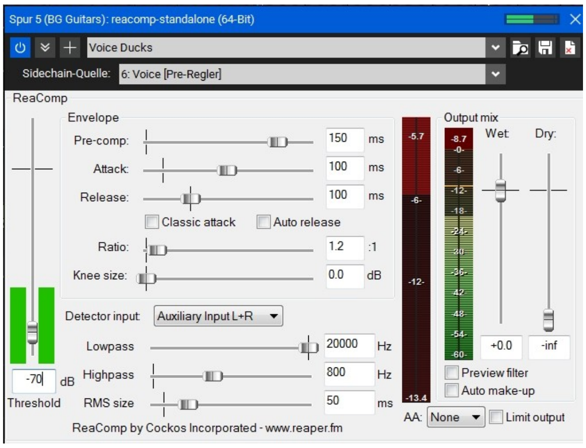
Not many DAWs provide a *ducker* plug-in, although a ducker can be eminently useful for tasks such as slightly turning the background guitar down during the vocals are on or during some solo instrument plays, which eliminates a lot of manual gain riding.

This subject does not strictly belong here, but it beautifully illustrates the *interconnection of threshold and ratio (as* explained in chapter 4.1). Deep compression can be achieved with very small (low) ratio values if the threshold is low enough. By setting the threshold very deep, all material is compressed evenly. This would normally be what **Izhaki** calls *static compression*, which (because it does not change) is nothing but a permanent volume reduction – exactly what we want.

Although technically speaking a ducker behaves different, it happens to be a happy byproduct of a compressor that it can produce a decent ducking effect using the side chain.

A very good demonstration of this is given in [Reapermania2015].

I have used the following settings with success:



I have gone a step further by setting the threshold *really low.* Together with some side chain EQ a really smooth compression / static volume reduction is achieved that persists as long as the side chain signal is active.

Note that I have chosen shorter timing values because they seemed too long for the purpose.
I felt that about 6dB of reduction was perfect. The very low ratio of 1.2 achieves that. The look-ahead helps a lot to anticipate the coming edge, a bigger RMS window makes the GR meter less nervous.

The same preset can be used basically to duck the bass guitar with the kick drum.

6 The Weapon Of Choice

In the preceding chapters we have managed to reduce our most basic compressor needs to situations that ask for *containing peaks, balancing levels* and *loudening,* which appear due to *micro dynamic* and *macro dynamic movements* throughout the song.

The seasoned pro sound engineer would not hesitate to grab one of the traditional units for those tasks, however, he or she would probably do so because they grew up with them and know them inside out – which certainly does *not* apply to the majority of us amateurs. There is a pretty well documented history of those applications: the **LA-2A** (its long name, *leveling amplifier* is self explanatory) for vocals [balancing levels], a **1176** for drums [containing peaks], **Fairchild** for mastering [loudening] and so on.

This is why we will try to roughly emulate them to gain a better understanding of the parameters that lend those attributes to them. Note that we do not attempt or claim to *exactly clone* them (there´s no shortage of high-priced specimen available that do this), but instead to *extract their basic quality*: in the example above, the *first one is slow, the second fast, the third subtle*, which we can use to get our above mentioned compression tasks done properly. Embellishments are not our primary concern at this point in time.

But what about modern (software) units, that do not even have a hardware ancestor? Should they not be much better in all respects?

> *"It's good to understand the parallels/inspiration for the [Fabfilter] C2 settings, but a better question may be: does your music NEED hardware-emulation compressors, or is something like C2 more than enough? There's no right or wrong answer."*
>
> - **Gearspace:** vitocorleone123 (thread # 7): Emulating Vintage Compressors in Pro C2, https://gearspace.com/board/music-computers/1262672-emulating-vintage-compressors-pro-c2.html

One of the currently most revered units is the **Fabfilter pro-C**. Tone Boosters have produced a take at this one with **compressor 4.** I tried the latter.

I found that they do not have all the parameters accessible we wish. Most of this is governed (and concealed) by their internal *all-knowing artificial intelligence algorithms.* While that may work (extremely well) for some situations, it again disempowers you. Do we know, if their "vocal" preset works for other balancing tasks? No. Does "aggressive 8:1" give a hint as to what is under the hood or what it may be useful for? No. This again leaves you in a weak state of not knowing really and subject a half hearted choice of presets.

That said, the above mentioned **pro C2** compressor has incredible visualizations that can be of great help. Some may object that *listening is everything*, and they may be right to an extent, but *hearing is of transitory nature, visuals have a certain persistence.* That is a clear advantage. Yet overall, this is no plausible alternative for our goals unfortunately.

The idea arose several years ago to use a single *compressor engine* and harness its parameters for those tasks rather than to have to resort to an endless list of specialized tools we know little about. Most times they hide their questionable innards behind fancy looks.

Reacomp gives us full access to all those parameters, so it appears like an ideal candidate. I did not any other one that does all that. We would assign model names according to the task, namely *leveling, containing peaks,* and *loudening,* which are unmistakable and universal in their usage rather than *optical* etc., which again give no clue for the intended task and introduce yet another level of abstraction and mystery.

For the exercise, let us look at some of those iconic units, their peculiarities and their preferable application. Most of the concepts are borrowed from [QuantumMusic2015-2] and are supplemented by me (*):

> *„"Should I buy a LA-2A or a 1176 ?". I mean, these are completely different machines. What are you trying to achieve in terms of texture exactly ?"*
> - Christopher A. Dion [QuantumMusic2015-2]

What we are trying to achieve has been worked out in the preceding chapters and should be clear by now. We are now about to choose the weapon that does it.

There are four big families of compressors. Let´s start with

6.1 VCA (Voltage Controlled Amplifier)

- Peak-sensing

- fast attack and release
- mostly hard-knee
- preferred usage: percussion (micro-dynamic), transients
- not recommended for: macro-dynamic
- can produce a sense of punch and aggressiveness
- can sound thin
- well-known representatives: SSL, Neve, API mix bus compressors, Focusrite RED, DBX 160, C1 (some of them transformer coupled, some not)

6.2 Opto

> *“In the digital world, the opto effect can be simulated using an “RMS” based compressor.”*
> - Christopher A. Dion [QuantumMusic2015-2]

- sensing an average that has been calculated over a time “window” (RMS)
- preferred usage: macro-dynamics
- not recommended for: peaks and transients (leads to pumping)
- caveat: bass heavy material will make it pump too (>side chain filtering)
- Very transparent. Leaves transients intact.
- well-known representatives: LA-3A, JLM Mac Opto Comp, LA-2A, TubeTech CL1B, * Avalon VT737SP
- * use look-ahead
- * note that there is no “one” LA-2A unit… The uaudio page says:

> *“Like the hardware, the LA-2A Silver, LA-2A Gray, and LA-2 models offer distinct variations in time constants, compression knee, headroom, distortion, program and frequency dependence, and more. ”*
> - https://www.uaudio.de/uad-plugins/compressors-limiters/teletronix-la-2a-collection.html

6.3 Variable Mu (Vari-Mu, Tube Compressor)

- Large span soft knee

- The actual ratio increases with gain reduction[19]. The louder a transient is, the harder it is going to be compressed.
- Very slow time constants.[20]
- preferred usage: mix glue
- not recommended for: ironing out dynamic issues or to get punch
- Very smooth compression
- warm, rich tone.
- Caveat: lack of SC filtering kills bass.
- well-known representatives: Fairchild 670, Altec 436C, Manley Variable Mu

6.4 FET (Field Effect Transistor)

> *“What Opto and variable mu simply can’t achieve, this one does. The slowest attack time available on the FET is usually faster than the fastest attack time on a variable mu! Yet, it’s far from being as transparent as a VCA. It definitely has more character.”*
> - Christopher A. Dion [QuantumMusic2015-2]

- preferred usage: drums, vocals, bass, and everything else that needs bite and punch
- not recommended for: mix without a SC, for transparency.
- rich and warm distortion (even without compression)
- * character compressor.
- well-known representative: 1176LN, *API 525
- * and yes, there are several models of this unit with distinctly different sonic properties too...

Christopher A. Dion’s home made mantra:

> *“Stop putting VCAs on everything simply because that’s the default compressor design that comes with every DAW.”*
> - Christopher A. Dion [QuantumMusic2015-2]

19 Note that that is the wording of the writer (C. A. Dion). As a matter of fact, the Ratio is fixed, it is just approached so gradual (by the huge knee span), that it appears like that, presumed you were putting a tangent to the curve at a given point.

20 Again, this is the way the writer (C. A. Dion) experienced it. As a matter of fact the Fairchild has a 200µs attack time, which is incredibly fast (faster than any of the contenders except FET), however if the unit uses a large RMS window (and there is much indication for that), all of that gets swamped in its slow behavior. The Manley’s attack time starts with 25ms, which is not necessarily slow either.

As we can see, the classical units' application fields are pretty well narrowed down.

But before attending to **Reacomp**, let us resolve the issue with its purported flaws.

6.5 Sidestep: Is Reacomp flawed?

As briefly mentioned above, **Dan Worrall** [Worrall2019] described in his video irregularities he found in **Reacomp.** Let's look at this closer.

- **the unit does not reach the targeted ratio values in feed-back mode.**

  Now this gets a bit technical. Imagine for the moment a feed-forward compressor. It takes its reference from the unaltered (uncompressed) input. The feed-back compressor in contrary, takes its reference from the output, that invariably *is* already compressed. Compressed is another word for "made smaller". So it has less voltage to compare to than the feed-forward type and thus less voltage into the gain control element. More ratio > more compression > even less voltage. Pretty fast this unit runs out of control voltage, which is why some professional units use *gain* in the side-chain. Also, the reference voltage has its corners rounded by attack and release time constants, so invariably, there will be some ratio related rounding-off of the ratio curve akin to a (variable) knee setting. Many modern compressors' *vocal* settings exhibit a type of "S" ratio curve, where ratio initially rises and then deteriorates to 1:1 again for transients. The acute reader will have noticed that we can set up our own side-chain and add gain to that if we wished.

So Worrall is correct in his observation of this peculiarity, but not in his conclusion that this is a bug, because this is a characteristic **all feed-back type compressors exhibit by design.** That said, **Reacomp** indeed does not go beyond a ratio of 2:1. I guess they implemented insufficient gain in the side chain.

> *"Notice that all feedback type compressors have a max reduction of about 30 dB. That is because you can never get a feedback-based device to compress with a ratio of more than 5:1. If you take the FB circuit and turn it into an FF circuit — optimize it for 20:1 ratio, then switch it back to FB mode — it will never get over 5:1 because of the gain riding FB does. (...)*
>
> *The units that go over 5:1 typically have gain in the control voltage. If you were to take this control design and turn a 20:1 FB circuit into an FF circuit, the ratio would be over inf:1 and would be doing negative reduction. (The dbx 160X had this feature.) (...)*
>
> *The sonically strange thing is that a 5:1 ratio in FB mode sounds more compressed than 5:1 should. That's because you can't accurately correlate ratios on FB compressors because they act differently with tones as compared to program material. It's a "dynamic" thang! (...)*
> - Paul Wolff (formerly of API) [Tangible2008]

> *[API 525)Another reason that that FB compressors were "limited" to 30 dB of Gain Reduction is that they could only attenuate as much as they were set up to lose."*
>
> - Paul Wolff (formerly of API) [Tangible2008]

This "dynamic" thing is exactly what we are after, and this is very likely what gives those units their distinct *vintage flavor*.
Worrall however is correct in his observation that the *make-up gain slider should not interfere with compression* and probably the observations about *oversampling*. They seem to have reacted to this according to [Worrall2019-2], but this feature does not seem to be incorporated in the stand alone version currently distributed (**Reaplugs** rev. 2.36 from 2016)[21].

This is easy remedied by leaving the slider in its default position and adding a simple gain plug-in afterwards, which is pretty painless. So no, **Reacomp is not flawed to a point of being unusable – not in this respect**. Conversely, **Worrall**'s stacked compressor work-around cannot be the remedy of a flaw that is none. It must do something different, since it can achieve higher ratio values. It suspect it will not be what a vintage unit does, but who knows...

I come to believe that **Reacomp** is not the be-all and end-all of compressor precision, but it is no doubt the most flexible one. And it is free. I would gladly use a commercial unit with comparable flexibility, if I found one.

21 I read in a forum that the plug-ins that come with **Reaper** are different from the ones that come in the free plug-in suite. Comprehensible in a way.

6.6 Models of prominent representatives of each hardware type

Let us now try to create models that roughly emulate the above characteristics using **Reacomp** (of course: *sans mojo embellishments)*

Note: from the insights we have gained so far, all of the following iconic units are *feed-back*. Read the *caveats* regarding using them below.

	SSL-G style	LA-2A style	1176 style	Fairchild style	DBX-160 style
Type	VCA	Opto	FET	Vari-Mu	VCA
Best used for	Punch, drums, bus	Leveling Vocals	Punch, Snare, Vocals	Mixbus, vocals,	See SSL
Char-acter	Aggressive, edgy, pumping	Glassy, polished	Bright, aggressive	smooth and warm in lows	See SSL
Task	containing levels	balancing levels	containing levels	loudening	containing levels
Precomp	0	0	0	0	0
Attack	**10ms** (0.1, 0.3, 1, 3, 10 and 30 ms)	Fixed @**10ms**	20 µs to 800 µs	.2/.2/.4/.8/.4/.2 ms (links to release) * see text	6ms
Release	**300ms** (0.1, 0.3, 0.6 and 1.2s, and Auto)	Fixed @40-**80ms** (*see text)	50 ms to 1.1s, program depen-dent, **use 600ms**	.3/.8/2/5/2/.3 sec **modern prefer-ence: short**	50ms
Auto Release	No	**try**	No	No	no
Ratio	2/4/20 (links to knee)	Fixed at **4**	4/8/12/20 (links to knee)	Fixed at **20** (see text)	4
Knee	2.5/1.5/0.5dB (est. from Cytomic manual) start with **1dB** *	> 0,9 (increasing with compression i.e. threshold) start with **1dB** *	2/1.5/1/0.5dB (from manual) start with **1dB** *	20dB **use: action range**	2dB
Detector Input	Output=Feed-back (*see text)	Output=Feedback (*see text)	Output=Feedback	Output=Feedback	feed forward (can do both)
RMS size	1.5ms	50ms (likely 140ms)	5ms	Likely low (*see text)	20ms
Embel-lishment	tr-saturation	tu/tr-saturation 6k high shelf	tr-saturation	tu/tr-saturation	

Caveat:

- Don’t forget to **leave the make-up slider alone** (leave at default position).
- Use a separate gain plugin (like GGain) for make-up if needed!

Using this table:

- “classical attack” does not imply some sort of “vintage” modeled attack, as I had assumed earlier. It refers to an older version of the plug-in. Best to ignore.
- The values given represent the result of my research. Boldfaced values are used for our initial settings. Note that some of the parameters may be wrong, since precise and hard facts are not easily found. Feel free to contact me if you know more.
- Options present in the **Reacomp** panel (like **Auto-Makeup, Output Limiter**) that are not addressed here are to be left **unchecked**.

- Side chain filtering has to be applied if necessary. This depends on the audio source. See chpt. 5.5.
- Embellishment options include tube/transformer saturation plug-ins in series
- Category “best used for”: applies for our task specific usage, but there is no hard and fast rules. Some people use unexpected units with success for a creative effect. If somebody uses, say an 1176 for vocals, he might see the need for containing the peaks but not for leveling. There is no way we can know that.

Remarks on the individual models:

- DBX160 model seen at [Worrall2019-1] at 17:30
- For La-2A release: 0.06 seconds for 50% release, 0.5 to 5 seconds for complete release depending upon the amount of previous reduction. **Reacomp**’s auto release has been found (by users) not to be a suitable substitute due to its unknown behavior.
- The LA-2A’s manual is contradictory. The block diagram and schematic clearly indicates a feed-forward compressor type (SC signal input-derived), while the text under “side chain circuit” claims a feed-back topology (SC signal output derived). Many other sources seem to support feed-back.
- The LA-2A’s user manual shows a graph that displays the connection between Ratio and Knee. The values are taken from that. From what we have learned so far we can attribute this dependency on the way feed-back mode influences the whole behavior. For a start, we may choose a small knee value (1dB) and with feed-back mode the model will exhibit the desired knee variation by design.
- The LA-2A has a heavily program dependent release. **Reacomp**’s *Auto Release* might work.
- The SSL-G is based on feed-back topology, but is very fast. To mimic a contemporary generic VCA style compressor, feed-forward is recommended. Technically, for fast “jobs” a feed-forward design (with look-ahead) is preferable, but the decision is not that straight-forward [SonicScoop-Slater2018].
- The SSL-G exhibits a similarly variable (with ratio) knee shape as the 1176 does. The Cytomic “The Glue” (plug-in sound-a-like) manual gives some clues to that. The values in the table are a first approximation.
- The Manley (as a proponent of vari-mu) manual does not specify any value for Ratio. This is clear if the knee is as large as it is. We don´t need to be fearful when choosing a high Ratio value due to that fact. Rate tops out at 20:1, and that is what we will use.
- Attack on the Manley ranges from 25-70ms.
- The Fairchild is much faster (200µs). It used to have the fastest attack of all before FETs came. It is uncertain where C. A. Dion (at the beginning of chapter 5.4) has his impression from that Vari-Mu’s are slow. He probably refers to the release time constant that can be chosen huge.
- For the same reason, it is unlikely that those units are RMS based to a noteworthy degree. This impression is further supported by the reference in Pendulum Audio’s *Quartet II* handbook (a modern evolution of the Fairchild) that clearly distinguishes between: *“Fast , Faster, Vintage (with program-dependent response), Average (rms responding), or full manual operation.”* And the Fairchild *is* fast. In fact it falls under the category *faster* by their definition, namely into the sub-millisecond range.

- What further supports this assumption is a look at the tube circuit [Tangible2008]. There is, apart from standard attack and release components, nothing that could function as an integrator (as different to an optical unit).
- That said, as requested in Chapter 5.3 (Loudening) we may still wish some RMS while keeping the other time constants low.
- For all models: attack and release timing delay values are chosen dependent on the task (see earlier chapters)
- LA-2A vs 1176: note that both are feed-back, and their main striking difference is their speed. For our purposes, all other things equal and given a ratio of 4, the RMS slider alone may bring us either into LA-2A or 1176 land.
- If a feed-back model can *truly* go in ratio noteworthy beyond 4, it must use additional circuitry *like side-chain gain or* switch to *feed-forward* upon higher ratios unbeknownst to us.
- This appears plausible considering that an additional tube gain stage (at the time of their introduction) was expensive, but a solid state counterpart was ludicrously cheap.

6.7 Task specific models

The following are “modern” (feed-forward) type implementations. This makes particularly sense where speed is of concern (> look-ahead).

	Balancing Levels	Containing Peaks	Loudening	N.Y. Stealth	N.Y. RMS
Best used for	Macro dynamics e.g. Vocals	Micro dynamics, transient shaping e.g. Snare,	Mastering, Mixbus, glue	parallel compression, e.g. vocals, bass,	
Character	Glassy, polished	Bright, aggressive	smooth and warm in lows	unobtrusive	to-the-face
Initial Threshold	Bottom of the action range	Bottom of the peak or top of action	set at the base of the whole signal	set at the base of the action	Set at center of action
Look ahead	0	20ms	-	20ms	0
Initial Attack	AF for LPI AS for PI (*see text)	AF	AF for LPI AS for PI	Min	Medium (125ms)
Initial Release	As above	AF		250-350ms	As above
Ratio	Start with 3	20 ?	High (50)	2.5	2 to 20 (see chpt. 5.12
Knee	1dB	<1	20 or action span	hard	Hard (small for higher ratios)
RMS size	50ms LPI Peak (5ms) PI	<5ms	Likely low (*see text)	peak	>100ms
SC HP filter	recommended	none			
Make-Up Gain	As needed	As needed	identical to threshold, but reverse in magni-tude.	This control takes on a different meaning: it controls the level of compressed signal that is added to the untreated signal	

- AF and AS refer to almost fastest and almost slowest. PI = percussive instruments like drums, LPI= less percussive instruments like vocals.
- Unmentioned parameters are off in **Reacomp**.
- Note that any *truly fast attack* will inevitably require some look-ahead.
- Use *feed-forward architecture* on all models.

We see that those models are not that far from our vintage models. Note the striking similarity of models “balancing levels” and “LA-2A style”. Naturally, *feed-back architecture* will instill some *vintage flavor,* so try to switch to feed-back mode and *behold*...

7 Final words

With **Reacomp**'s reputation adequately restored (for ratio values < 2:1 at least), we can easily cycle through the vintage models and behold their beautiful quirks. They can serve as an excellent foundation. The emulation models were created with great care based upon a lot of research. The sources thereof are so numerous that they have to wait for publication until some other day. Besides that, not many of them are *official*.

However, the *task oriented models* can easily be implemented *as is*. They will most certainly be more "perfect" than the vintage ones in many respects. It turns out, that what distinguishes them the most is **RMS size, knee size** and where the side chain derives its **reference signal** from (input or output).
Feed-back mode will by design force a compression curve that exhibits deterioration of ratio ("S"-curve for a lack of better words) besides being responsible for interactions between the parameters such as a variable knee.
Switching to feed-back is a sure ticket to a vintage compressor flavor and can be used with **Reacomp** with minor restrictions.

As mentioned initially, the models depicted are meant to fix level problems early in the mixing process, so that we can instill some sense of loudness into our tracks for a rough mix.

I believe that they are an excellent starting point together in conjunction with the strategies given in earlier chapters. Note that we have barely touched upon parameters like *attack* and *release*, but this kind of fine tuning can wait for a point in time later down the mixing process.

There also hides a myriad of other powerful tricks in compressors, but this is a can of worms that would exceed the dimensions and the intent of this paper.

I hope this article helped you to choose the right *parameters* for a certain task with authority rather than the right *plug-in,* which could save you a lot of grief, time and likely money.
Or conversely, it may help you to use the plug-ins you have with more authority.

Let me know how you get on with the models depicted.

8 Disclaimer

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9 Reference

[Massey2009]
H. Massey, *Behind the glass* (Vol II). Backbeat Books.
[Senior2012]
M. Senior, *Mixing Secrets*, 1st ed (German), mitp.de
[Izhaki2012]
R. Izhaki: *Mixing Audio: Concepts, Practices and Tools (2nd ed.)*, Oxford M.A., Elsevier, 2012
[Stavrou2003]
M.P. Stavrou, *Mixing with your mind*, Reprint 2015, Mullumbimby, Australia, Flux Research Pty Ltd.
[Katz2010]
B. Katz: orig. title *Mastering Audio-the art and the science (*2nd ed), German ed., Munich 2010, GC Carstensen Verlag

---

All subsequent documents can be found in the WWW

[Reardon2013] link obsolete
B. Reardon, *Compression: Compressors in Use,* http://bartreardon.ipower.com/www/Compression.html
[Tarr2014]
E. Tarr for ProAudioFiles: *5 Effective Ways to Use Compression,* https://theproaudiofiles.com/compression-techniques/
[Cochrane2016-1]
G. Cochrane (http://therecordingrevolution.com/): *How to simplify compression into two easy steps*, (email to subscribers)
[ProAudioFiles2016]
The Pro Audio Files; *The Basics of Attack Time on a Compressor,* https://www.youtube.com/watch?v=yACwEyGjgiE&index=38&list=PLe1lxu8fHox_fycpNskW1Po6k2Qsih9Uf
[SOS2013]
Hugh Robjohns for SOS magazine**,** *Parallel Compression - The Real Benefits,* http://www.soundonsound.com/techniques/parallel-compression

[TalkingMusic]
Use parallel compression to enhance sounds**,** https://talkinmusic.com/musicproduction/use-parallel-compression-to-enhance-sounds/
[Cochrane2016-2]
G. Cochrane (video): *Mixing With EQ - The Smart Volume Knob,* https://www.youtube.com/watch?v=cI3diHXO7r8
[QuantumMusic2015]
C. A. Dion, *Advanced Compression Techniques - Compression in-depth: The beginning,* http://quantum-music.ca/wordpress/index.php/category/advanced-compression-techniques/
[QuantumMusic2015-2]
C. A. Dion, VCA, Opto, Vari-Mu, *FET compressors… When to use which ?,* http://www.quantum-music.ca/wordpress/index.php/2015/07/29/vca-opto-varimu-fet-compressors-when-to-use-which/
[SOS-Senior2009]
Mike Senior: *Classic Compressors: Choosing The Right Compressor For The Job,* https://www.soundonsound.com/techniques/classic-compressors
[SonicScoop-Slater2018]
Sonic Scoop / Rick Slater: *Feedback Vs. Feed-Forward Compression: The Differences You Need to Know,* https://sonicscoop.com/feedback-vs-feed-forward-compression-differences-need-know/
[Worrall2019-1]
Dan Worrall: *What's Wrong With Stock Plugins? ReaComp;* *https://www.youtube.com/watch?v=FpXqYk1FoWA*
[Worrall2019-2]
Dan Worrall: *What's Wrong With Stock Plugins? ReaComp ReaVisited;* *https://www.youtube.com/watch?v=7Yit769SN64*
[Tangible2008]
Tangible Technology (Paul Wolff, formerly of API): *An Overview of Compressor / Limiters and their Guts;* *https://tangible-technology.com/dynamics/comp_lim_ec_dh_pw2.html*
[Sknoteaudio]
Sknote Audio: *Why a Sidechain High Pass Filter in a Compressor Makes Sense;* *https://www.sknoteaudio.com/wp/index.php/2015/12/16/sidechain-high-pass-filter-compressor/*
[Reapermania2015]
Reapermania: *Ducking using Side-Chain Compression in REAPER (ReaComp);* *https://www.youtube.com/watch?app=desktop&v=8Tt5yWOxB2E*

Many thanks to Martin Pyka, who offered beautiful style sheets for **Open Office** free of charge on his website:http://www.martinpyka.de/2005/03/23/openoffice-vorlagen-3/ The application of such style sheets is easy. Documents formatted according to professional typographic rules are far more readable and enjoyable.