



**NONLINEARLABS**

# C15



**Sound Generation Tutorial**



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Document Version: 1.9

Date: September 21, 2023

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## About these tutorials

These tutorials were written to make you find your way into the secrets of your C15 synthesizer quickly and easily. Please consult the Quickstart Guide or the User Manual to learn everything about the basic concept and setup of your C15 before using these tutorials. Please also consult the User Manual any time to delve deeper into the capabilities of the C15 synthesis engine, and to learn about all the details of any of the parameters of the instrument.

A tutorial will teach you basic aspects of the C15's concepts as well as the various components of the sound engine, and how they interact with each other, in a hands-on manner. It is an easy way to familiarize yourself with your C15, and a starting point for your sound design work on the instrument it is, too.

If you feel like learning more about the details of a specific parameter (e.g. value ranges, scaling, modulation capabilities etc), please refer to chapter 8.4. "Parameter Reference" of the User Manual at any time. You can use the tutorials and the User Manual in parallel.

The tutorials use the front panel of the instrument. In case you would rather like to work with the Graphic User Interface, please refer to the Quickstart Guide or chapter 7 "User Interfaces" of the User Manual first to learn about the basic concepts of the GUI. After this, you will be able to easily apply the programming steps described and transfer them from the hardware panel to the GUI.

### Formats

These tutorials describe fairly simple programming examples you can follow step by step. You will find lists boasting programming steps and figures that show the C15's user interface state. To make things perfectly clear, we make use of specific formatting throughout the entire tutorial.

Buttons (Section) that need to be pressed are formatted in bold print. The name of the section follows in (brackets). The Encoder is labelled the same way:

**Sustain** (Envelope A) ... **Encoder** ...

Secondary parameters that can be accessed by repeatedly hitting a button are labelled in bold italic: ***Asym***

Data values are bold and in square brackets: [ **60.0 %** ]

Controllers, as the Ribbons and Pedals, are labelled in Bold Capitals: **PEDAL 1**

Programming steps to be performed are indented to the right and marked with a triangle, like this: ▶

Notes on the previous programming step are even more indented to the right and marked with a double slash: //

This will look e.g. like this:

### Applying modulation to the PM self modulation of Oscillator A:

- ▶ Press **PM A** (Oscillator B) two times. **Env A** is highlighted in the display.
- ▶ Turn the **Encoder** to [ **30.0 %** ].  
// Oscillator B is now being phase-modulated by the signal of Oscillator A.  
The modulation depth is controlled by Envelope A at a value of 30.0%.

Every once in a while, you will find some notes of particular importance (at least we believe so...). They are marked by an exclamation mark (which looks like this:

⚠ Please note that there is ...

Sometimes, you will find some explanations within a list of programming steps. They provide a little more in-depth knowledge and are called "Excursions". They look like this:

#### **e** *Excursion: Parameter Value Resolution*

Some parameters need a ...

Here and there, you will find short recaps – they look like this:

#### **Recap: Oscillator section**

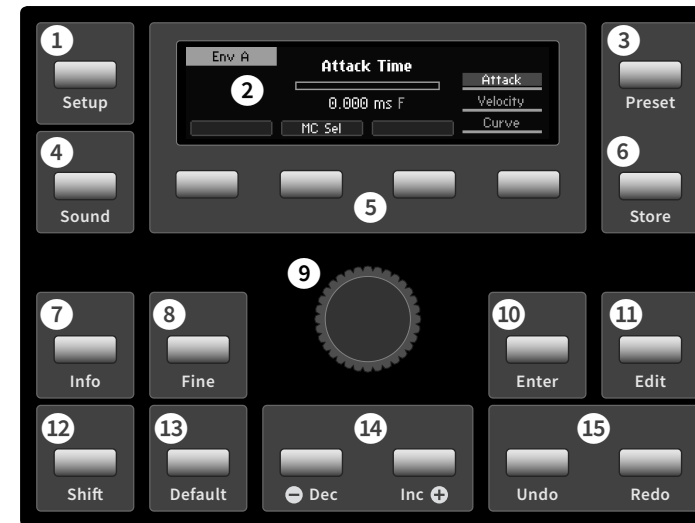
## Basic Conventions

Before starting, it is crucial to understand some basic conventions of the front panel – more on this in the Quickstart Guide:

- When a button on a Selection Panel is pressed, the parameter is selected and its value can be edited. Its LED will light up permanently. Additional “Sub Parameters” can be accessed by pressing the button multiple times.
- There can be some flashing LEDs to show the targets of the signal that is generated in the selected Parameter Group.
- When a Macro Control is selected, flashing LEDs display the parameters it is modulating.
- When the Preset screen is on, the currently active signal flow or active parameters respectively are indicated by LEDs lighting up permanently.

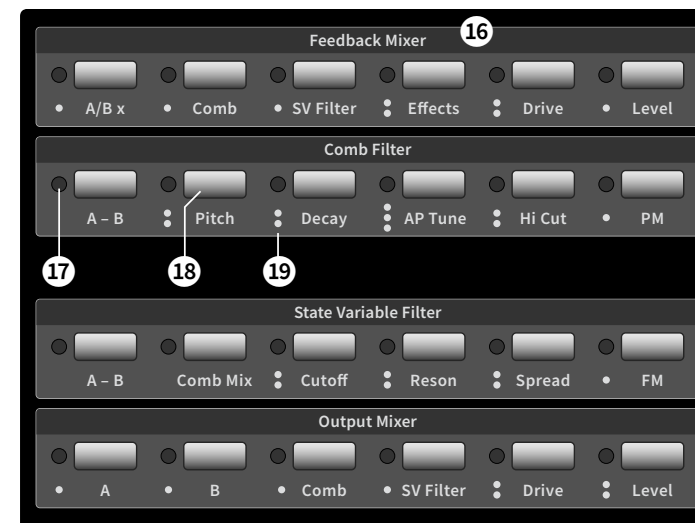
## Hardware User Interface

The images on the next page show the Edit Panel and one of the Selection Panels of the Panel Unit, and the Control Panel of the Base Unit.



### Edit Panel

- 1 Setup Button
- 2 Panel Unit Display
- 3 Setup Button
- 4 Sound Button
- 5 Soft Buttons 1 to 4
- 6 Store Button
- 7 Info Button
- 8 Fine Button
- 9 Encoder
- 10 Enter Button
- 11 Edit Button
- 12 Shift Button
- 13 Default Button
- 14 Dec / Inc Buttons
- 15 Undo / Redo Buttons



### Selection Panel

- 16 Parameter Group
- 17 Parameter Indicator
- 18 Parameter Selection Button
- 19 Indicators for Sub Parameters



### Base Unit Control Panel

- 20 +/- Buttons
- 21 Base Unit Display
- 22 Funct / Mode Buttons

The first tutorial describes the basic functions of the sound generation modules, their interaction (resp. modulation capabilities), and the signal path. You will learn how to create specific waveforms using the oscillators, blend them, and feed them into subsequent modules like filters and effects. We will be dealing with the filters as sound-processing devices as well as with the sound-generating capabilities of the Comb Filter. The tutorial will be topped off by an insight into the feedback capabilities (which is another very interesting way of creating sounds).

As you certainly know already, the C15's oscillators initially generate sine-waves. The real fun starts when these sine-waves are warped to generate complex waveforms with amazing sonic results. We will start right there:

## Init Sound

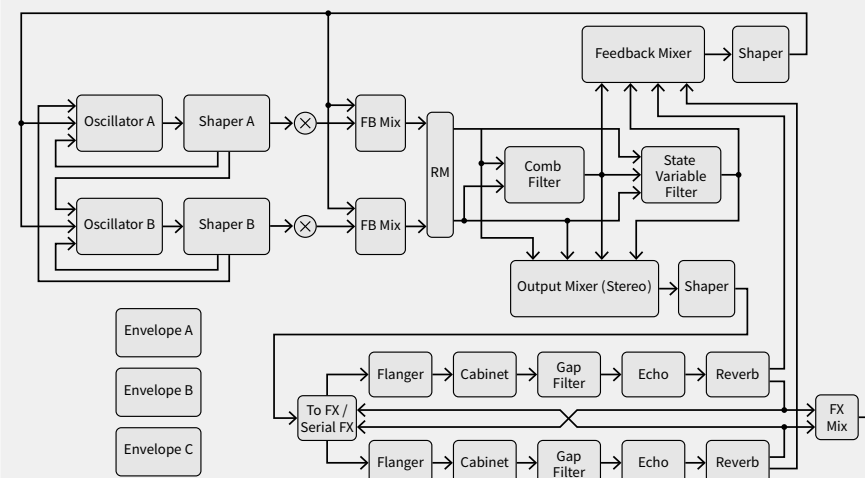
Starting with the Init Sound is the best thing to do. When loading the Init Sound, parameters are set to their default values (same thing happens when using the default button). The Init Sound uses the most basic signal path with no modulations at all. Most of the mix parameters are set to zero value.

### Initializing all parameters (resp. the edit buffer):

- ▶ Press **Sound** (Edit Panel).
- ▶ Press and hold **Default** (Edit Panel).
- ▶ Now you can choose whether you want to initialize the edit buffer as a Single, Layer or Split Sound (Edit Panel > Soft Button 1-3).  
// Now the edit buffer has been initialized. You will hear – nothing. Don't worry, you are not the one to blame. Please proceed:
- ▶ Press **A** (Output Mixer).
- ▶ Turn the Encoder to approx. [ **60.0 %** ].
- ▶ Play some notes.  
// You will hear the typical Init sound – a simple, slowly decaying one-oscillator sine-wave sound.

### Excursion – A Short Glimpse at the Signal Path

Before we proceed further, let's have a brief look at the C15's structure / signal path:



Starting point are the two oscillators. They produce sine-waves for a start but these sine-waves can be warped in various ways to produce complex waveshapes. This is done by phase modulation (PM) and by using the Shaper sections.

Each oscillator can be phase-modulated by three sources: itself, the other oscillator, and the feedback signal. All three sources can be used at the same time in variable proportions.

Three Envelopes control both Oscillators and Shapers (Env A → Osc/Shaper A, Env B → Osc/Shaper B, while Env C can be routed pretty flexibly, e.g. for controlling the filters).

To process the oscillator signals even further, there is a State Variable Filter as well as a Comb Filter. When operating at high resonance settings and being pinged by an oscillator signal, both filters can work as signal generators in their own right.

Oscillator/Shaper outputs and filter outputs are fed into the Output Mixer. This section allows you to blend and balance various sonic components with each other. To avoid undesired distortion at the output stage, keep an eye on the Output Mixers Level parameter. Values around – 4.5 or – 5 dB are mostly on the safe side. If you want to use distortion deliberately to produce timbral variations, please consider using the Drive parameter of the Output Mixer or the Cabinet effect instead.

The final stage of the signal path is the effects Section. It is fed from the Output Mixer where all voices are combined into a monophonic signal. When using the Init sound, all five effects will be bypassed.

## Oscillator Section / Creating Waveforms

A typical parameter screen of the Panel Unit display looks like this:



- |                  |                       |                           |
|------------------|-----------------------|---------------------------|
| 1 Group Header   | 3 Graphical Indicator | 5 Soft Button Labels      |
| 2 Parameter Name | 4 Parameter Value     | 6 Main and Sub Parameters |

### Oscillator Basics

#### Let's (de)tune Oscillator A:

- ▶ Press **Pitch** (Oscillator A)
- ▶ **A-B** (Comb Filter) **A-B** (State Variable Filter) and **A** (Output Mixer) are flashing to show you that both filters and the Output Mixer are receiving a signal from the selected Oscillator A (even though you are not hearing much filtering right now).
- ▶ Turn the **Encoder** and detune Oscillator A by semitones.  
// The pitch is displayed in MIDI-note numbers: "60" is MIDI note 60 and equal to note "C3". It is the pitch that you hear when playing the third "C" of the keyboard.

#### Now let's play around with Key Tracking:

- ▶ Press **Pitch** (Oscillator A) two times. Its light stays on. Now watch the display. It shows the highlighted parameter **Key Trk**.  
// Note that multiple hitting of a parameter button toggles between the upper "main" parameter (here "Pitch") and several "sub" parameters (here **Env C** and **Key Trk**) that are related to the main parameter.
- ▶ Turn the Encoder to [ **50.00 %** ]. Oscillator A's keyboard tracking is now halved which equals playing quarter-tones on the keyboard.

- ▶ Turn the Encoder to [ **0.00 %** ]. Each key is playing at the same pitch now.  
// A key tracking close to 0.00% can be very useful when an oscillator is used as LFO-like modulation source or slow PM-carrier.  
More on this later. . .
- ▶ Turn the Encoder back to [ **100.00 %** ] (the usual semi-tone scaling).  
// Reset every parameter to its default value by hitting **Default** (Edit Panel).

#### Let's introduce some envelope parameters:

(please consult the User Manual for all details of the envelope parameters or use the Info button on the Edit Panel).

- ▶ Press **Attack** (Envelope A).
- ▶ Turn the **Encoder** and play some notes.
- ▶ Press **Release** (Envelope A).
- ▶ Turn the **Encoder** and play some notes.  
// Envelope A is always connected to Oscillator A and controls its volume.
- ▶ Press **Sustain** (Envelope A).
- ▶ Turn the **Encoder** to approx. [ **60,0 %** ].  
// Oscillator A is now providing a static signal level.

### Oscillator Self-Modulation

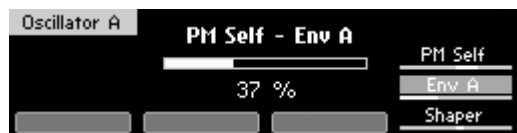
- ▶ Press **PM Self** (Oscillator A).
- ▶ Turn the **Encoder** back and forth.  
// Oscillator A's output is fed back into its input. At higher rates, the output wave gets increasingly warped and generates a sawtooth wave with rich harmonic content. Sweeping the Encoder will produce a filter-like effect.

#### **e** Excursion – bipolar parameter values

PM Self works at positive as well as negative parameter values. You will find many more parameters with positive and negative values, not only modulation depth settings (as you might know from other synthesizers) but also mixing levels etc. In many cases, a negative value represents a phase-shifted signal. Only when mixing such a signal with other signals, phase cancellations will generate audible effects. With Self PM active, a positive value will generate a sawtooth-wave with rising edge, negative values generate a falling edge.

## Let's make Oscillator self-modulation dynamic and control Self-PM of Oscillator A by Envelope A:

- ▶ Set the **Encoder** to approx. [ 70,0 % ] self modulation amount.
- ▶ Press **PM Self** (Oscillator A) again. Watch the Display: **Env A** is highlighted  
// You have just accessed the first sub-parameter "behind" PM-Self ("Env A"). It is the amount of Envelope A modulating PM-Self of Oscillator A.



- // Alternatively, you can toggle through the sub-parameters behind the currently active button with the rightmost soft button at any time.
- ▶ Turn the **Encoder** to [ 100,0 % ].  
// Envelope A now provides a dynamic modulation depth for PM Self of Osc A. As a result, you will hear a transition from bright to soft or the other way round, depending on the settings of Env A.
- ▶ Now tweak the different Envelope A parameters a bit (see above): Depending on the settings, you will hear some simple brassy or percussive sounds.  
// Since Envelope A is influenced by keyboard velocity, the sound will also depend on how hard you are hitting the keys.

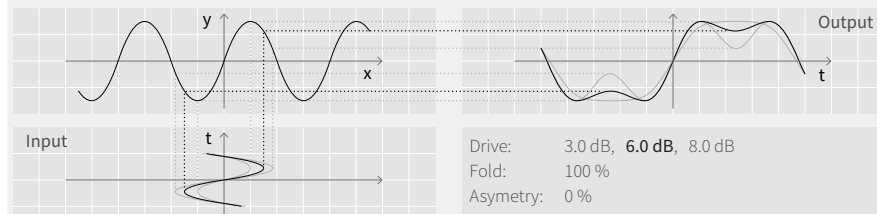
## Introducing the Shaper

- ▶ First, please reset Oscillator A to a simple sine-wave by selecting **PM Self** and **PM Self - Env A** (Env A) and hitting **Default**. Envelope A should be providing a simple organ-like setting.
- ▶ Press **Mix** (Shaper A).
- ▶ Turn the **Encoder** slowly to [ 100.0 % ] and play some notes.  
// At increasing Mix values, you will hear the sound getting brighter. Note that the sound is somewhat different from the results of "PM Self".  
Now the Oscillator A signal is being routed through Shaper A. "Mix" blends between the pure oscillator signal (0 %) and the output of the Shaper (100 %).
- ▶ Press **Drive** (Shaper A).
- ▶ Turn the **Encoder** slowly and play some notes.

- ▶ Then set the Drive to [ 20.0 dB ].
- ▶ Press **Fold** (Shaper A).
- ▶ Turn the **Encoder** slowly and play some notes.
- ▶ Press **Asym** (Shaper A).
- ▶ Turn the **Encoder** slowly and play some notes.  
// Fold, Drive and Asym(metry) warp the signal to generate various waveshapes with very different harmonic content and timbral results.
- ▶ Press **PM Self** (Oscillator A) again.
- ▶ Turn the **Encoder** to [ 50.0 % ] and play some notes.
- ▶ Press **PM Self** (Oscillator A) again.
- ▶ Turn the **Encoder** slowly and play some notes.  
// Now you have just fed the Shaper with the self-modulated (resp. sawtooth wave) signal instead of a sine wave.

### e Excursion – what's that Shaper doing?

In simple words, the Shaper distorts the oscillator signal in various ways. It maps the input signal to a shaping curve to produce a more complex waveform. Depending on the settings, a vast range of different harmonic spectra can be created.



The **Drive** parameter controls the intensity of distortion induced by the Shaper and can produce a vaguely filter-like effect.

The **Fold** parameter controls the amount of ripples in the waveform. It emphasizes some odd harmonics while the fundamental is attenuated. The sound gets some characteristic "nasal" quality, not unlike a resonating filter.

**Asymmetry** treats the upper and the lower part of the input signal differently and generates even harmonics (2nd, 4th, 6th etc) that way. At high values, the signal is pitched one octave higher while the fundamental is eliminated.

All three parameters interact with each other, producing countless variations of distortion curves and resulting waveforms.



**e** *Excursion – the C15's signal routing / blending*

As with all signal routings in the C15, the Shaper is not switched in or out of the signal path but continuously blended with another (usually the dry) signal. This makes sense since it provides great morphing capabilities without any steps or clicks in the sound. More on this later.

**e** *Excursion – parameter value fine resolution*

Some parameters need a very fine resolution to fine-tune a sound the way you desire. In order to do this, the resolution of every parameter can be multiplied by a factor of 10 (sometimes even 100). Simply hit the **Fine** button to toggle fine resolution on and off. To get an impression of that effect, try out “**Drive** (Shaper A)” in fine resolution mode.

By selecting a new parameter, the fine “mode” will automatically be disabled. To enable fine resolution permanently, press **Shift + Fine**.

- ▶ Now set **PM Self** to [ **75 %** ].
- ▶ Press **PM Self** (Oscillator A) another two times (or use the **rightmost soft button**) to access the sub-parameter **Shaper**. It is highlighted in the display.
- ▶ Turn the **Encoder** slowly and play some notes.  
// Now the signal for the phase-modulation of Oscillator A is fed back post Shaper: Instead of a sine-wave, a complex waveform is now used as modulator. This generates even more overtones and, beyond a certain degree, it can produce increasingly chaotic results, noisy or “chirpy” sounds in particular. You will hear the effect of the shaper even when you set the shaper's Mix parameter to zero.

**Both Oscillators together****Mixing both Oscillators:**

- ▶ First, please reload the Init Sound. Both Oscillators are now generating simple sine-waves again.
- ▶ Press **A** (Output Mixer).
- ▶ Turn the **Encoder** to approx. [ **60.0 %** ].
- ▶ Press **B** (Output Mixer).

- ▶ Turn the **Encoder** to approx. [ **60.0 %** ].  
// Now, both oscillators are sending their signals through the Output Mixer.
- ▶ Press **Level** (Output Mixer).
- ▶ Turn the **Encoder** to approx. [ **-10.0 dB** ].  
// You have just reduced the mixer's output signal enough to avoid unwanted distortion.
- ▶ Press **Sustain** (Envelope A).
- ▶ Turn the **Encoder** to [ **50 %** ].  
// Oscillator A is now providing a sine-wave at a constant level whereas Oscillator B is still fading out over time.

**Creating intervals:**

- ▶ Press **Pitch** (Oscillator B).
- ▶ Turn the **Encoder** to [ **67.00 st** ]. Play some notes.  
// Now Oscillator B is tuned seven semitones (a fifth) above Oscillator A. You may also try out different intervals like e. g. an octave (“72”) or an octave plus an additional fifth (“79”).
- ▶ Turn the **Encoder** back to [ **60.00 st** ] or use the Default button.
- ▶ Press **PM Self** (Oscillator B).
- ▶ Turn the **Encoder** to approx. [ **60.0 %** ]. Play some notes.  
// Oscillator B is modulating itself now, sounding brighter than Oscillator A.
- ▶ Press **Decay 2** (Envelope B).
- ▶ Turn the **Encoder** to approx. [ **300 ms** ].  
// Oscillator B is now fading out at a medium decay rate. The resulting sound is vaguely reminiscent of a piano of sorts.
- ▶ Press **Sustain** (Envelope B).
- ▶ Turn the **Encoder** to [ **50%** ].  
// Now, both Oscillators are producing steady tones. The resulting sound is vaguely reminiscent of an organ.

You have just created some sounds that are made up of two components: A basic sine-wave from Oscillator A and some sustained / decaying overtones from Oscillator B. Very simple still, but with lots of creative options to choose from ...

## Detuning Oscillator B:

- ▶ Press **PM Self** (Oscillator A).
- ▶ Turn the **Encoder** to [ **60.00 %** ].  
// We simply wanted to make the entire sound somewhat brighter, to improve the audibility of the following example.
- ▶ Press **Pitch** (Oscillator B).
- ▶ Press **Fine** (Edit Panel).
- ▶ Sweep the **Encoder** slowly up and down and dial in [ **60.07 st** ].  
// Oscillator B has now been detuned by 7 Cents above Oscillator A. Detuning generates a beat frequency that we all love so much because it makes the sound so “fat” and “vibrant”.

## Tweaking the sound a bit more:

- ▶ Press **Attack** (Envelope A and B).
- ▶ Turn the **Encoder**.
- ▶ Press **Release** (Envelope A and B).
- ▶ Turn the **Encoder**.  
// Adjust PM Self level and Envelope parameters as you please. Depending on the settings, the results will vary between string and brass-like sounds.

## The same beat frequency at all pitch ranges with Key Tracking

As you may have noticed, the beat frequency changes across the range of the keyboard. Higher up the keyboard, the effect can grow too strong and sound a bit “unnatural”. To achieve a steady beat frequency at all pitch ranges:

- ▶ Press **Pitch** (Oscillator B) three times. **Key Trk** is highlighted in the display.
- ▶ Press **Fine** (Edit Panel).
- ▶ Turn the **Encoder** slowly to [ **99.80 %** ].  
// At a Key Tracking below 100%, the pitch of higher notes will be increasingly reduced resp. not proportional to their position on the keyboard. This detunes high notes a bit less than low notes and keeps the beat frequency lower in high ranges, resp. steady across a wide pitch range.

## One oscillator modulating the other:

- ▶ First, please reload the Init-Sound. Don’t forget to turn up **Level A** on the Output Mixer to [ **60.0 %** ]. Both Oscillators are now generating simple sine-waves. What you are hearing right now is Oscillator A.
- ▶ Press **PM B** (Oscillator A).
- ▶ Turn the **Encoder** and dial in approx. [ **75.00 %** ].  
// Oscillator B is not added to the output mixer but used to modulate the phase of Oscillator A instead. Since Oscillator B is currently generating a sine-wave at the same pitch as Oscillator A, the audible effect is similar to self-modulation of Oscillator A. But here comes the fun part, we are now detuning Oscillator B:
- ▶ Press **Pitch** (Oscillator B).
- ▶ Sweep the **Encoder** and play some notes. Then dial in [ **53.00 st** ].  
// You will now be hearing some soft “metallic” timbres that sound quite promising (but that’s only us, of course...).

### **e** Excursion – The Secrets of Phase Modulation (PM) – Oscillator Pitches and Modulation Index

When modulating the phase of one oscillator by another one at a different frequency, lots of sidebands or new overtones respectively are generated. Those were not present in the source signals. The frequency ratio of both oscillator signals defines the harmonic content resp. the overtone structure of the resulting signal. The resulting sound remains harmonic as long as the ratio between the modulated oscillator (called “carrier” – here Oscillator A) and the modulating oscillator (called “modulator” – here Oscillator B) is a proper multiple (1:1, 1:2, 1:3 etc). If not, the resulting sound will become increasingly inharmonic and dissonant. Depending on the frequency ratio, the sonic character is reminiscent of “wood”, “metal” or “glass”. This is because the frequencies in a vibrating piece of wood, metal or glass are very similar to the frequencies generated by PM. Obviously, PM is a very good tool to generate sounds that feature this type of timbral character. A second crucial parameter is the intensity of the phase modulation or “modulation index”. In the C15, the appropriate parameters are called “PM A” and “PM B”. Different values will produce radically different timbral results. The interaction between the pitch of the respective oscillators and their modulation depth settings (“PM A / B”) is also crucial to the sonic results.

## Controlling the Modulator by an Envelope:

---

As you have learned in the meantime, frequency and mod depth of the modulator (here Oscillator B) are crucial for shaping sound using PM. Unlike classic subtractive synthesis, it is very easy to generate a wide range of noisy and “metallic” timbres that offer lots of potential when emulating acoustic instruments, like e.g. mallets or plucked strings. To explore this, we will now add some sort of percussive “stroke” to a simple sound:

- ▶ Load the Init sound and turn up Oscillator A (the carrier):  
**A** (Output Mixer) = [ **75.0 %** ]
- ▶ Press **Pitch** (Oscillator B).
- ▶ Set the **Encoder** to [ **96.00 st** ].
- ▶ Press **PM B** (Oscillator A).
- ▶ Set the **Encoder** to approx [ **60.00 %** ].  
 // Now you are hearing Oscillator A being phase-modulated by Oscillator B.  
 The sound is bright and slowly decaying.
- ▶ Press **Pitch** (Oscillator B) until **Key Trk** is highlighted in the display.
- ▶ Turn the **Encoder** and dial in [ **0.00 %** ].  
 // The Key Tracking of Oscillator B is off now, providing a steady modulator-pitch for all keys. In some key ranges, the sound is now becoming somewhat odd.
- ▶ Press **PM B** (Oscillator A) until **Env B** is highlighted in the display.
- ▶ Set the **Encoder** to [ **100.0 %** ].  
 // Now Envelope B is controlling the phase-modulation depth (PM B) over time.
- ▶ Press **Decay 1** (Envelope B).
- ▶ Turn the **Encoder** to [ **10.0 ms** ].
- ▶ Press **Decay 2** (Envelope B).
- ▶ Turn the **Encoder** to approx. [ **40.0 ms** ] and play some notes. Keep Breakpoint (BP Level) at default value 50%.  
 // Envelope B is now producing a short percussive “stroke” that quickly fades out. In every key range, the percussive “stroke” sounds slightly different since the pitch ratio between carrier and modulator is a bit different for every key. This helps to make emulations of natural sounds pretty realistic.

## Using Key Tracking as a sound parameter:

---

- ▶ Press **Pitch** (Oscillator B) until **Key Trk** is highlighted in the display.
- ▶ Turn the **Encoder** and dial in [ **50.00 %** ] while playing some notes.  
 // The Key Tracking of Oscillator B has been enabled again which forces Oscillator B to change its pitch depending on the note played. As you remember, pitch ratios between oscillators are altered and hence the harmonic structure of the resulting sound will also be altered across the entire note range. Enjoy trying out some timbral results.

## Using the Modulator Pitch to change the sonic character:

---

- ▶ Now change **Pitch** (Oscillator B).  
 // You will notice the timbral transition from “wooden” (medium pitch ranges) through “metallic” to “glassy” (high pitch ranges).  
 Re-adjust **Decay 2** (Envelope B) a bit also and you will hear some simple but amazing “tuned percussion” sounds.  
 As a pretty nice-sounding example, dial in e.g. **Pitch** (Oscillator B) 105.00 st and **Decay 2** (Envelope B) 500 ms. Have fun and get carried away (but not too much) ...

## Cross modulation:

---

- ▶ Press **PM A** (Oscillator B).
- ▶ Turn the **Encoder** slowly up and dial in approx. [ **50.00 %** ].  
 // The phase of Oscillator B is now being modulated by Oscillator A. That means, both oscillators are now modulating each other’s phase. This is called cross- or x-modulation. That way, lots of inharmonic overtones are produced and, accordingly, the sonic results can be quite odd and often noisy. They depend very much on the frequency/pitch ratios of either oscillators (please see above). Please feel free to explore some nice Pitch B values and Envelope B settings as well as variations of PM A and PM B and the modulation of PM A by Envelope A. At proper parameter value ratios, you may create some nice “plucked strings” – nylon and steel strings included.

### **e** Excursion – Adjusting velocity sensitivity

You certainly want to explore a lot of expressive potential when enjoying your sounds. The C15 provides a lot of capabilities to do so (Ribbon Controllers, Pedals etc). For starters, we would like to introduce Keyboard Velocity. Its default setting is 30.0 dB which works pretty well in many cases.

- ▶ Press **Level Vel** (Envelope A).
- ▶ Turn the **Encoder** and dial in [ **0.0 dB** ] first, then increase the value slowly to [ **60.0 dB** ] while playing some notes.
- ▶ Repeat the process with Envelope B.  
// Since Envelope A controls the level of Oscillator A, a change of its velocity value affects the loudness of the current sound. Oscillator B level (the Modulator) is controlled by Envelope B. Since Oscillator B determines the timbral character of the current setting to some extent, its level has a huge effect on the current sound.

### **Oscillator as LFO (Low Frequency Oscillator):**

Now set up your C15 so that

- Oscillator A produces a steady sine-wave (no Self-PM, no Envelope modulation)
- Oscillator A is constantly phase-modulated by Oscillator B (again no Self-PM, no Envelope modulation here). **PM B** (Oscillator A) should have a value around [ **90.0 %** ] to make all the following sonic results easily audible. Oscillator B should not be part of the audible output signal, i. e. B (Output Mixer) is [ **0.0 %** ].
  - ▶ Press **Pitch** (Oscillator B).
  - ▶ Sweep the **Encoder** up and down while playing some notes.  
Then dial in [ **0.00 st** ].  
// You will hear a fast pitch vibrato. Its frequency is depending on the note played.
  - ▶ Press **Pitch** (Oscillator B) until **Key Trk** is highlighted in the display.
  - ▶ Turn the **Encoder** and dial in [ **0.00 %** ].  
// The Key Tracking of Oscillator B is set to Off now which results in a constant pitch (and vibrato speed) across the entire note range.

Now Oscillator B is behaving like an (almost) ordinary LFO and can be used as a source for periodic modulation in the sub-audio range. Please note that, in contrast to most other (analog) synthesizers with a dedicated LFO, the C15 sports an oscillator/LFO per voice. They are not phase-synced which helps to animate many sounds in a natural way.

### Recap: Oscillator section

The C15's combination of two oscillators and two shapers, controlled by two envelopes, allows the generation of many different kinds of waveshapes from simple to complex:

- Initially, both Oscillators produce sine-waves (without any overtones)
- With Self PM active, each Oscillator generates a variable sawtooth wave (with all overtones)
- When routed through the Shaper, depending on the settings of Drive and Fold, various rectangle and pulse-like waveforms can be generated (with odd-numbered overtones).
- The Shaper's Asym(metry) parameter adds even harmonics.  
⚠ Interaction of the parameters mentioned above produces a wide timbral scope and dramatic timbral shifts.
- Mixing both Oscillator/Shaper outputs in the Output Mixer produces sounds with two sonic components, as well as intervals and out-of-tune effects.
- Phase Modulation (PM A / PM B) of one Oscillator by the other as well as cross-modulation can produce inharmonic sounds. The pitch ratios of the Oscillators and the modulation settings mainly determine the timbral results.  
⚠ Careful adjustment of pitch, Key Tracking and mod depth settings is important for timbre as well as for making pitched sounds playable! Use Fine resolution to adjust the crucial parameters.
- Introduction of Envelope A and B produces dynamic control over level and timbre.
- Oscillators can be used as LFOs when key tracking is disabled.

## The State Variable Filter

To introduce the State Variable Filter (S.V. Filter), we should first set up the oscillator section to produce a sawtooth waveform which is rich in overtones. This is a good input signal fodder to explore the State Variable Filter.

First, please load the Init sound – this time, you do not need to crank up “A” on the Output Mixer!

- Set Oscillator A’s PM Self to 90 % for a nice sounding saw-wave.
- Set Envelope A’s Sustain to 60 % in order to produce a steady tone.

Now please proceed like this:

### Enabling the S.V. Filter:

- ▶ Press **S.V. Filter** (Output Mixer).
- ▶ Set the **Encoder** to approx. [ 50.0 % ].  
// The “S.V. Filter” input of the Output Mixer is fully open now and you can hear the signal passing the filter. Since input “A” is closed, all you are hearing is the plain S.V. Filter signal.
- ▶ Press **A – B** (State Variable Filter).  
// This parameter determines the ratio between the Oscillator/Shaper signals A and B, fed into the S.V. Filter input. For now, keep it at its default setting “A”, i. e. [ 0.0 % ].

### The very basic Parameters:

- ▶ Press **Cutoff** (State Variable Filter).  
// S.V. Filter (Output Mixer) is flashing to let you know that the S.V. Filter is part of the signal path.
- ▶ Sweep the **Encoder** across the entire value range and dial in the default value [ 80.0 st ].  
// You will hear the characteristic transition from bright to dull since overtones are gradually being eliminated from the signal.  
! At very low settings, when the cutoff setting is below the frequency of the fundamental note, the output signal might become inaudible.
- ▶ Press **Reson** (State Variable Filter).

- ▶ Sweep the **Encoder** across the entire value range and dial in the default value [ 50.0 st ].  
// When increasing resonance values, you will hear frequencies around the cutoff setting becoming increasingly edgier and more pronounced. Cutoff and resonance are the most effective filter parameters.

### e Excursion – Controlling the current Parameter using Ribbon 1

Sometimes, it can be more useful (or funnier) to control a parameter using a ribbon controller rather than an encoder. This is useful when performing with the parameter as well as adjusting values very accurately.

To assign a Ribbon to a specific parameter (here the Cutoff of the S.V. Filter), simply:

- ▶ Press **Cutoff** (State Variable Filter).
- ▶ Press **Mode** (Base Unit Control Panel) until the Base Unit Display shows **Cutoff**. This mode is also called Edit Mode.
- ▶ Slide your finger across **RIBBON 1**.  
// The parameter currently selected (Cutoff) is now controlled by **RIBBON 1**, or your finger tip



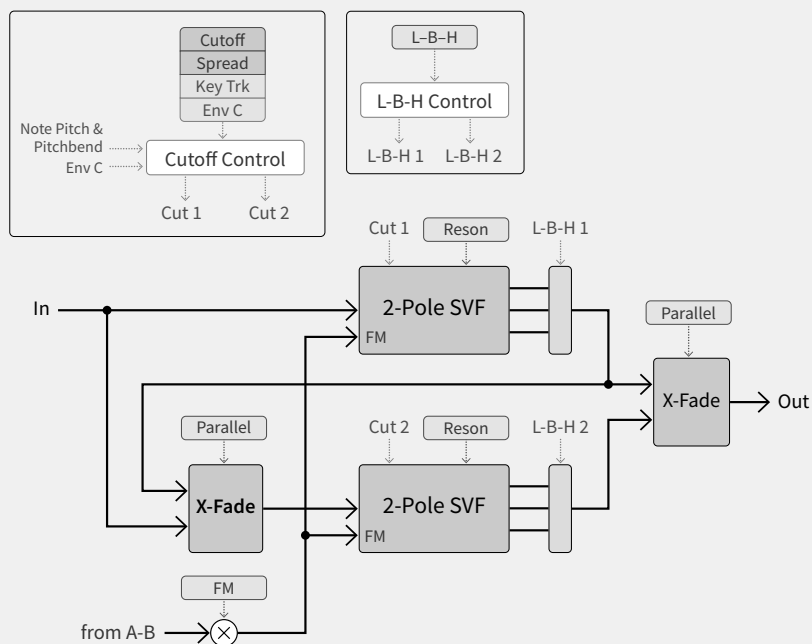
⚠ When using the C15’s Macro Controls, the Ribbons/Pedals can control various parameters at the same time. This very interesting topic will be covered in a later tutorial. Stay tuned.

### Exploring some of the more advanced S.V. Filter parameters:

⚠ Our word of advice: No matter whether you are familiar with filters in general or not, please grab the user manual and take some time to study all those flashy S.V. Filter parameters in detail.

## e Excursion: The S.V. Filter functionality

The S.V. Filter is a combination of two resonating two-pole state-variable filters, each with a slope of 12 dB. Cutoff and Resonance can be controlled manually or modulated by Envelope C and Key Tracking.



The spacing between both cutoff-points is variable (“Spread”). The filter characteristics can be swept continuously from low through band to high-pass mode (“L-B-H”). Both filters work in series by default but can be continuously shifted to parallel operation (“Parallel”).

- Setting Spread to 0.0 st creates a simple four-pole filter. At higher Spread values, the spacing between the two Cutoff frequencies increases.
- Cutoff and Resonance always affect both filter sections in the same manner.
- L-B-H determines the characteristics of both filter sections:
  - L – both filter sections work in lowpass mode. High frequencies are attenuated, producing a sound that can be described as “round”, “soft”, “fat”, “dull” etc.
  - H – both filter sections work in highpass mode. Low frequencies are attenuated, producing a sound that can be described as “sharp”, “thin”, “bright” etc.

- B – the first filter section works as a highpass, the second as a lowpass. Low and high frequencies are both attenuated and a frequency band with variable width (“Spread”) passes the S.V. Filter. Particularly at higher Resonance settings, vowel/vocal-like sounds can be achieved.
- FM provides a Cutoff modulation by the Oscillator/Shaper signals A and B. Very good for aggressive and distorted sounds.

Check out the parameters mentioned above and keep in mind that they all interact with each other in some way. Use the Default button to reset a parameter value.

### Envelope / Key Tracking modulation of Cutoff and Resonance:

- ▶ Press **Cutoff** (State Variable Filter) until **Env C** is highlighted in the display .
- ▶ Set the **Encoder** to [ 70.00 st ].
  - // You will hear the sound becoming increasingly dull over time since the Cutoff is modulated by Envelope C.
  - Vary the settings of Envelope C parameters and the modulation depth (“Env C”). For more dramatic filter “sweeps” set the Resonance of the S.V. Filter to higher values.
- ▶ Press **Cutoff** (State Variable Filter) until **Key Trk** is highlighted in the display.
- ▶ Sweep the **Encoder** across the entire range and dial in [ 50.0 % ].
  - // When set to 0.0 %, Cutoff has the same value across the entire keyboard range. When reducing the Key Tracking value, the Cutoff value will increase in higher keyboard ranges and the sound grows brighter – an effect you can find with many acoustic instruments.
  - Please check Env C / Key Trk modulation of the Resonance as well.

### Changing the Filter Characteristics:

- // The S.V. Filter is a four-pole filter composed of two two-pole filters. The Spread parameter determines the interval between the two cutoff frequencies of these two parts.
- ▶ Set the **Resonance** to [ 80 % ].
- ▶ Press **Spread** (State Variable Filter).
- ▶ By default, Spread is set to 12 semitones. Try settings between 0 and 60 semitones and also vary the Cutoff.
  - // When reducing the Spread value, the two peaks will emphasize each other and the result will be a very intensely resonating, “peaking” sound.

- ▶ Press **Spread** (State Variable Filter) again until L-B-H is highlighted in the display.
- ▶ Sweep the **Encoder** across the entire value range and dial in the default value [ **0.0 %** ] (Lowpass).  
// Using the L-B-H parameter, you can morph continuously from lowpass through bandpass to highpass. 0.0 % is fully lowpass, 100.0 % fully highpass. The width of the bandpass is determined by the Spread parameter.

### Cutoff FM:

- ▶ Press **FM** (State Variable Filter).
- ▶ Sweep the **Encoder** across the entire range.  
// Now the filter input signal is modulating the Cutoff frequency. Usually, the sound gets increasingly nasty and abrasive. Please note that positive and negative FM can produce quite different results.
- ▶ Press **FM** (State Variable Filter) until **A – B** is highlighted in the display.  
// A – B blends between the Oscillator/Shaper signals A and B and determines the signal ratio which is modulating the Filter Cutoff. Depending on the waveshape and pitch of both Oscillator/Shaper signals, the results can differ considerably from each other.
- ▶ Reset **FM** and **A – B** to their default values.

## The Output Mixer

You have already laid your hands on the Output Mixer. Here you will find some more info on that module.

If you are only popping in at this point, we should first set the oscillator section to produce a sawtooth waveform:

- ▶ First, please load the Init sound – do not forget to crank up “A” on the Output Mixer!
- ▶ Set Oscillator A’s **PM Self** to [ **90 %** ] for a nice-sounding sawtooth-wave.
- ▶ Set Envelope A’s **Sustain** to [ **60 %** ] in order to produce a steady tone.

Now carry on, please:

### Using the Output Mixer:

- ▶ Press **S.V. Filter** (Output Mixer).
- ▶ Set the **Encoder** to approx. [ **50.0 %** ].
- ▶ Press **A** (Output Mixer).
- ▶ Set the **Encoder** to approx. [ **50.0 %** ].  
// You have just combined the output signal of the S.V. Filter with the direct (unfiltered) signal of Oscillator A.
- ▶ Sweep the **Encoder** across the entire value range and back to [ **50.0 %** ].  
// Positive Level values add signals. Negative Level values subtract the signal from the others.  
⚠ Because of phase cancellation, positive and negative values may produce different timbral results here and there. It is worth trying out both polarities of the Levels.  
⚠ Please note that high input levels can produce audible saturation effects that make the sound edgier and/or more aggressive. To avoid unwanted distortion in the subsequent stages (e.g. effect section), please compensate for the gain boost by reducing the output level of the mixer by using Level (Output Mixer).

### The Drive Parameter:

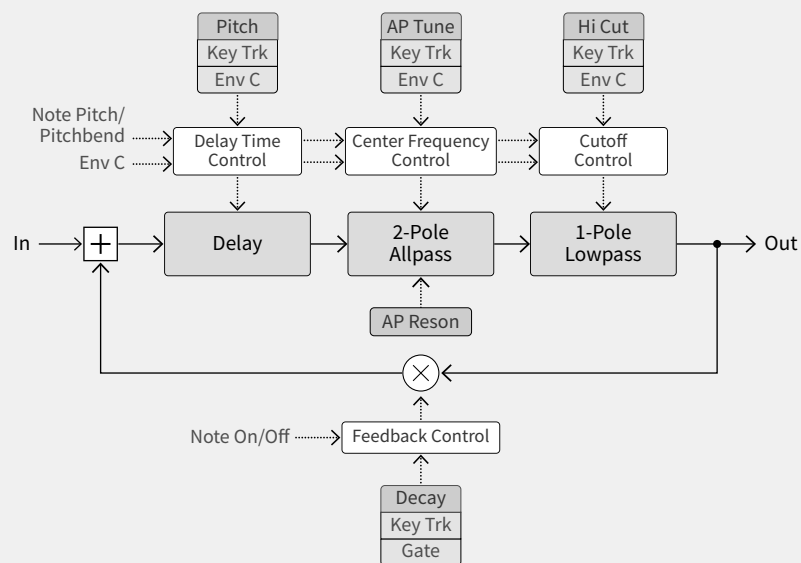
- ▶ Press **Drive** (Output Mixer).
- ▶ Sweep the **Encoder** across the entire value range.  
// Now the output signal of the mixer is passing through a flexible distortion circuit that produces everything from mild fuzzy distortion up to wildest sound mangling.  
Check out the Drive parameters **Fold** and **Asymmetry** as well. To avoid unwanted distortion in the subsequent stages (e.g. effect section), please compensate for the gain boost by reducing the output level of the mixer by using **Level** (Output Mixer).
- ▶ Reset all **Drive** parameters to their default values.

## The Comb Filter

The Comb Filter can shape an incoming sound by imposing specific characteristics on it. The Comb Filter can also work as a resonator and it can produce periodic waveforms like an oscillator this way. It is an integral part of the C15's sound generation, and it can be useful when achieving imbral characteristics of e.g. plucked or bowed strings, blown reeds, horns, and many strange things in between and far beyond that.

### Excursion – Comb Filter Basics

Let's have a brief look at the C15's Comb Filter structure:



Basically, a comb filter is a delay with a feedback path. Incoming signals pass the delay section and a certain amount of the signal is then being fed back into the input. The signals making their rounds in this feedback loop generate a tone that can be controlled by various parameters to achieve specific sonic characteristics and a dedicated pitch – the comb filter is turned into a resonator / sound source.

### Enabling the Comb Filter:

To explore the Comb Filter, dial in a simple sawtooth-wave sound – we have absolutely no reason to believe you don't already know how to do this. Okay, here comes a brief reminder for your convenience:

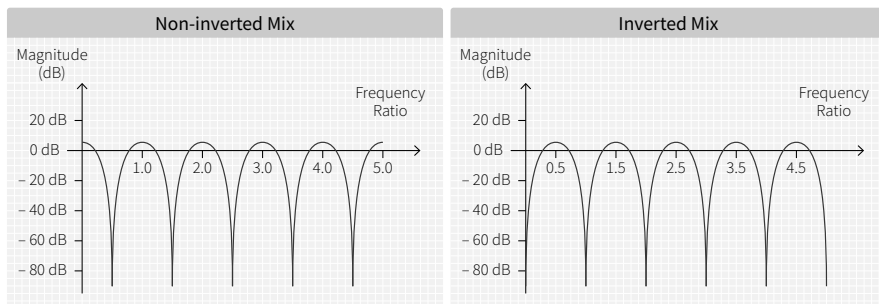
- ▶ Load the Init sound and set the Output Mixer level **A** to [ 50.0 % ].
- ▶ Press **Sustain** (Envelope A).
- ▶ Set the Encoder to approx. [ 80.0 % ].
- ▶ Press **PM Self** (Oscillator A).
- ▶ Set the **Encoder** to [ 90.0 % ].  
// Oscillator A is now generating a sustained sawtooth-wave.
- ▶ Press **Comb** (Output Mixer).
- ▶ Set the **Encoder** to approx. [ 50.0 % ].  
// The Comb Filter signal is now blended with the oscillator signal.
- ▶ Press **A – B** (Comb Filter).  
// This parameter determines the ratio between the Oscillator/Shaper signals A and B, fed into the Comb Filter input. For the time being, please keep it at its default setting “A”, i. e. 0.0 %.

### The very basic Parameters

#### Pitch:

- ▶ Press **Pitch** (Comb Filter).
- ▶ Sweep the **Encoder** slowly across the entire range and dial in [ 90.00 st ]. Please also try to control it by **RIBBON 1** in Edit Mode (please refer to page 25).  
// You will hear the sound change while turning the Encoder. The Pitch parameter is actually the delay time which is converted and displayed in semitones. The shifting sound coloration is a result of boosting or eliminating specific frequencies when the delayed signal is combined with the non-delayed signal. Please also try a negative value for one of the mixing levels.





### Decay:

- ▶ Press **Decay** (Comb Filter).
- ▶ Sweep the **Encoder** slowly across the entire range.
- ▶ Change both Pitch and Decay and try out the various timbral effects.

// Decay controls the feedback of the delay. It determines the amount of the signal making its rounds in the feedback loop, and thus the time it takes for the oscillating feedback loop to fade out. This depends very much on the delay time dialed in (“Pitch”). When changing Pitch slowly, you can hear the “peaks” and “troughs” in the frequency spectrum, i. e. boosted and attenuated frequencies.

⚠ Please note there are positive and negative Decay values. Negative values invert the phase of the signal (negative feedback) and provide different sonic results with a certain “hollow” character – good for e.g. bell-like timbres ...

### Exciting the Comb Filter:

So far, we have been working with a sustained / static input signal. Even more interesting is the use of an impulse to stimulate the feedback loop of the Comb Filter:

- ▶ Turn the output signal of Oscillator/Shaper A into a short and sharp “click” by dialing in suitable parameter values for Envelope A:

<b>Attack:</b>	0.000 ms	<b>Decay 1:</b>	2.0 ms
<b>Breakpoint:</b>	100%	<b>Decay 2:</b>	4.0 ms
<b>Sustain:</b>	0.0 %	<b>Release:</b>	4.0 ms

- ▶ Set **Decay** (Comb Filter) to [ **1000 ms** ]
- ▶ Set **Pitch** (Comb Filter) to [ **0.00 st** ] and slowly turn up the **Encoder** value while playing some notes. Then dial in [ **60.00 st** ].  
// At the lower end of the Pitch range, you will notice audible “reflections” of the delay line. Their number depends on the Decay setting (resp. the feedback level). At higher pitches, resp. shorter delay times, the reflections grow increasingly dense until they sound like a static tone that has a dedicated pitch.

### **e** Excursion – Some Nuts and Bolts of Physical Modelling

What you have just programmed into your C15 is a very simple example of a sound-generation type usually referred to as “Physical Modelling”. It comprises a dedicated signal source – the exciter – and a resonator, in our case the Comb Filter. The exciter signal stimulates the resonator, generating a “ringing tone”. Matching sympathetic frequencies of exciter and resonator are boosted, others attenuated. Depending on the pitch of the exciter (Oscillator pitch) and the resonator (delay time of the Comb Filter), these frequencies can vary a lot. The audible pitch is determined by the resonator. This method is characteristic of many acoustic instruments, e.g. a plucked string or a blown flute stimulating a resonating body of sorts.

### More advanced Parameters / Refining the Sound

#### Key Tracking:

- ▶ Press **Decay** (Comb Filter) until **Key Trk** is highlighted in the display.
- ▶ Sweep the **Encoder** across the entire range and dial in approx. [ **50.0 %** ].  
// Now, the Decay at higher note ranges is reduced, compared to lower note ranges. This produces a more “natural feel”, useful for many sounds which are to resemble specific acoustic qualities.

## Hi Cut:

---

- ▶ Press **Hi Cut** (Comb Filter).
- ▶ Sweep the **Encoder** across the entire range and play notes. Then dial in a value of [ **110.00 st** ].  
// The signal path of the Comb Filter features a lowpass filter that attenuates high frequencies. At maximum value (140.00 st), the lowpass will be opened completely with no frequencies attenuated, yielding a very bright sound. Decreasing the value gradually, the lowpass is producing an increasingly muffled sound with quickly decaying treble frequencies. These settings are very useful for emulating e.g. plucked strings.

## Gate:

---

- ▶ Press **Decay** (Comb Filter) until **Gate** is highlighted in the display.
- ▶ Sweep the Encoder across the entire range. Play some notes and dial in [ **60.0 %** ].  
// This parameter controls to what extent a gate signal reduces the Decay time of the Comb Filter as soon as a key is released. When disabled (0.0 %), the Decay will be the same throughout, no matter whether a key is depressed or released. Particularly in combination with Key Tracking, this also allows for very natural-sounding results, e. g. think of the behaviour of a piano keyboard.

## AP Tune:

---

- ▶ Press **AP Tune** (Comb Filter).
- ▶ Slowly sweep the **Encoder** from its maximum to its minimum value while repeating the middle “C” on the keyboard. Then dial in [ **100.0 st** ].  
// This parameter enables an allpass filter in the signal path of the Comb Filter. Usually (without the allpass filter), the delay time is the same for all passing frequencies. All the overtones generated (resp. their multiples) fit perfectly into the delay time range dialled in. But within resonant bodies of acoustic instruments, things are a bit more complex since the delay times change with frequency. This effect is emulated by the allpass filter. The overtones generated by the feedback loop are detuned against each other by the allpass which produces specific inharmonic sonic components. The lower the allpass filter is tuned, the more overtones are affected, and the timbral variations increase. This effect is audible e. g. in

the lowest octave of a piano, which sounds quite metallic. This is because the physical qualities of those heavy-gauge piano strings, found in the lowest octave, are quite similar to those of metal tines or plates.

- ▶ Press **AP Tune** (Comb Filter) until **AP Reson** is highlighted in the display.
- ▶ Sweep the **Encoder** across the entire range while playing some notes. Then dial in approx. [ **50.0 %** ].  
// The resonance parameter of the allpass filter adds a lot of sound-sculpting potential. Explore the interaction between **AP Tune** and **AP Reson** carefully. They produce approximations of sonic characteristics that are similar to metal tines, plates, and more.
- ▶ Reset all **AP Tune** parameters to their default values.

## Varying the Exciter Settings (Oscillator A)

Even when the Oscillator signal is not audible, its qualities are crucial to the resulting sound. Envelope shape, pitch, and overtone structure of the exciter have a profound impact on the resonator (Comb Filter).

### Envelope shape:

---

- ▶ Press **Sustain** (Envelope A).
- ▶ Set the **Encoder** to approx. [ **30.0 %** ]
- ▶ Press **Attack** (Envelope A).
- ▶ Set the **Encoder** to [ **100 ms** ]
- ▶ Press **Decay 2** (Envelope A).
- ▶ Set the value to [ **100 ms** ] (default).  
// Oscillator A – the exciter of the Comb Filter – will no longer provide a short ping but a steady tone.
- ▶ Press **Pitch** (Oscillator A).
- ▶ Sweep the **Encoder** slowly across the entire range and play notes. Then dial in [ **48.00 st** ]. Enjoy...  
// Depending on Oscillator 1 Pitch, you will find interesting resonating frequencies as well as frequency cancellations. The sonic character is sometimes reminiscent of (over)blown reeds or bowed strings.

## Using “Fluctuation”:

- ▶ Press **Fluct** (Oscillator A).
- ▶ Sweep the **Encoder** slowly across the entire range while playing some notes. Then dial in approx. [ **60.0 %** ].

// At various pitch ratios between Oscillator A (exciter) and Comb Filter (resonator), the frequency boosts and attenuations are very strong and limited to narrow frequency bands. Consequently, the peaks and notches are quite difficult to handle, and often it is difficult to achieve musically useful results, e. g. a steady tonal quality across a wide key range.

The Fluctuation parameter is a welcome aid at this point: It randomly varies the oscillator pitch and thus produces broader frequency bands with matching ratios. The peaks and notches are evened out, and the sound is becoming more consistent. The sonic character also changes – in our example, it is shifting from a reed instrument towards a string orchestra.

### Recap: Using the Comb Filter as a resonator

- The Comb Filter is a delay line with feedback loop, driven into oscillation and thus generating a tone.
- The Pitch parameter of the Comb Filter determines the delay time and thus the pitch of the generated tone.
- Frequency boosts and cancellations in the feedback loop create a complex frequency response that determines the timbral character.
- The Decay parameter controls the feedback amount and, by that, the number of repetitions of the input signal. This determines the decay time of the tone generated by the resonator.
- An oscillator signal (exciter) stimulates a response of the comb filter (resonator).
- The qualities of the exciter determine the timbral character of the resulting sound to a large extent.
- Short, percussive exciter signals produce sounds like plucked strings. Sustained exciter signals produce sounds like bowed strings or (over)blown woodwinds.
- Key Tracking and a Gate (on Decay) as well as a lowpass filter (“Hi Cut”) produce natural sounding characteristics of “plucked strings”.
- An allpass filter (“AP Tune”) can shift the overtones and provide sonic characteristics of “metal tines” or “metal plates”.

Listen to Oscillator A (the exciter) and the Comb Filter (the resonator) separately by changing the Output Mixer settings. The oscillator is currently producing a steady noise with a very broad frequency range. The Comb Filter “picks” its resonant frequencies and boosts them. Hence, the frequency ratio between exciter and resonator is crucial to the resulting sound. Parameters like the exciter’s volume envelope settings and all Comb Filter parameters also shape the sound and interact with each other. That way, the C15’s physical-modelling features will provide you with a vast field for timbral exploration.

## Using Feedback Paths

As you already know (at least we are confident you do), the C15’s signal path provides various ways of feeding back signals which means that certain amounts of signals can be tapped at a specific point in the signal flow and reinserted at an earlier stage. We will now explore how to create sounds by using these feedback structures.

First, please reload the well known Init sound. If necessary, please find a detailed description on page 10.

Second, dial in a typical Comb Filter sound with the character of a plucked string. This will require

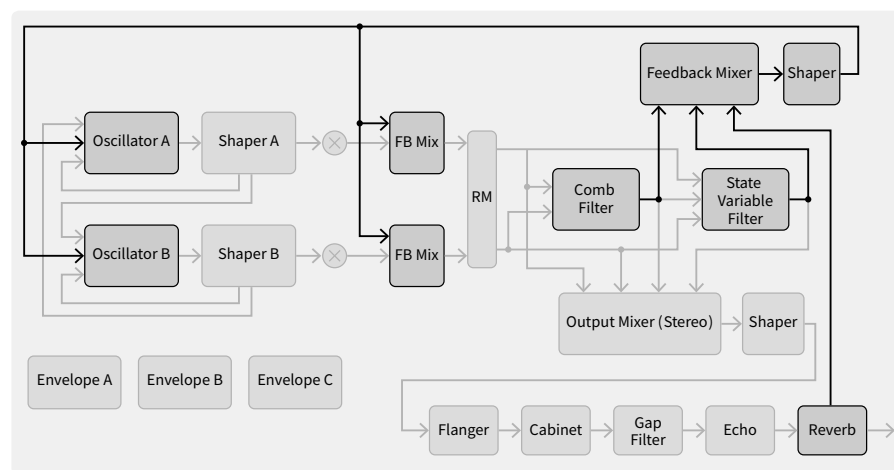
- the Comb Filter being mixed to the output (**Comb** (Output Mixer) around 50 %)
- a short exciter signal, resp. a very fast decaying oscillator sound (Envelope A: **Decay 1** around 1 ms, **Decay 2** around 5 ms) with plenty of overtones (high value for **PM Self**). It provides the “plucked” signal part which stimulates the comb filter.
- a comb filter setting with medium **Decay** time (around 1200 ms) and **Hi Cut** setting (e.g. 120.00 st). Set the Decay **Gate** to approx. 40.0 %.

If necessary, tailor the parameters a bit to your liking until the C15 sounds somewhat like a harpsichord. Now we are ready to proceed.

## Setting up a feedback path:

As mentioned earlier, sustained comb filter sounds can be achieved by continuous excitation of the comb filter (resonator). This can be done by using sustained oscillator signals. Another way to continuously excite the resonator is feeding a certain amount of its output signal back to its input. On the C15, this can be done by using the Feedback Mixer, which will be introduced right now:

- ▶ Press **Comb** (Feedback Mixer).
- ▶ Turn the **Encoder** to [ **40.0 %** ].  
 // By doing so, a certain amount of the Comb Filter's output signal is routed back to the Feedback bus. It could also be combined with the output signals of the State Variable Filter and the effects section. To fully enable the feedback path, the destination of the feedback signal needs to be determined. Available destinations can be found in the Oscillator and Shaper sections. We will use the "FB Mix" insert point located after the Shaper in the signal path. Please refer to the synth engine overview when you feel lost at this point.



- ▶ Press **FB Mix** (Shaper A).
- ▶ Turn the **Encoder** to [ **20.0 %** ]. Now you can hear sustained notes.  
 // The Comb Filter signal is tapped and routed back to the Comb Filter input as an exciter signal through the Feedback Mixer and feedback bus. If the loop gain is greater than 1, it will keep the filter constantly "ringing" with self-oscillation.

## Shaping the feedback sound:

... by using negative feedback level settings:

- ▶ Press **Comb** (Feedback Mixer).
- ▶ Turn the **Encoder** to [ **-40.0 %** ].  
 // At negative settings, the feedback signal is inverted. This will typically have a "damping" effect and shortens the sound produced. If you are operating the Comb Filter at negative Decay values, the negative values in the Feedback Mixer will drive it into self-oscillation.
- ▶ Press **Decay** (Comb Filter).
- ▶ Turn the Encoder to [ **-1260.0 ms** ].

... by applying the signal-shaping parameters of the Feedback Mixer:

- ▶ Press **Drive** (Feedback Mixer).
- ▶ Sweep the **Encoder** across the entire range.
- ▶ Press **Drive** (Feedback Mixer) again to access the parameters **Fold** and **Asymmetry**.
- ▶ Again sweep the **Encoder** across the entire range.  
 // As with the Output Mixer, the Feedback Mixer has a shaper stage that can distort the signal. The saturation of this stage limits the feedback level to avoid uncontrolled nastiness. Shaper curves allow a certain sonic control over the self-oscillating signal. Try out the effects of "Drive", "Fold", and "Asymmetry" and listen closely to the sonic results. Feedback level and polarity as well as the Drive parameters interact with each other.

... by modifying the Envelope / Oscillator A settings (exciter):

Still, the entire audible sound is generated by the comb filter only. Oscillator A is producing nothing but a short exciter signal which influences the resulting waveforms at the output of the Comb Filter – but is not audible itself. A lot of timbral variations can be achieved by adjusting the parameters of Oscillator A and its Envelope A.

- ▶ Reset the Parameters of **Drive** (Feedback Mixer) by using the Default button
- ▶ Press **Pitch** (Oscillator A).
- ▶ Sweep the **Encoder** across its entire range while playing notes and dial in [ **72.00 st** ].
- ▶ Press **Sustain** (Envelope A).

- ▶ Try different Sustain levels while playing notes and dial in approx. [ 5% ].
- ▶ Press **Fluct** (Oscillator A).
- ▶ Try different Fluctuation levels while playing notes.
  - // By changing the envelope, pitch, and signal spectrum of Oscillator A, the self-oscillating Comb-Filter will generate a plethora of different timbres. Please try longer Attack and Decay times as well as different settings of PM, Self, and the Feedback Mixer and FB Mix parameters.

### ... by filtering the feedback signal using the State Variable Filter:

First, let's return to a well-defined (and well-known) setting:

- ▶ Recall the **Init** sound.
- ▶ Set **Comb** (Output Mixer) to [ 50% ].
- ▶ Set **Decay 1** (Envelope A) to 1 ms and **Decay 2** (Envelope A) to [ 5 ms ].
- ▶ Set **PM Self** to [ 75% ].
- ▶ Set **Decay** (Comb Filter) to [ 1260 ms ] and **Hi Cut** to [ 120.00 st ].

Now we are creating a special feedback routing:

- ▶ Press **Comb Mix** (State Variable Filter).
- ▶ Turn the **Encoder** to [ 100.0% ].
- ▶ Press **S.V. Filter** (Feedback Mixer).
- ▶ Turn the **Encoder** to [ 50.0% ].
- ▶ Press **FB Mix** (Oscillator A).
- ▶ Turn the **Encoder** to [ 25.0% ].
  - // The State Variable Filter is now placed within the feedback path and is processing the signal arriving from the Comb Filter.
- ▶ Press **Spread** (State Variable Filter) until [ L - B - H ] is enabled.
- ▶ Turn the **Encoder** to [ 50.0% ] to enable the bandpass setting.
- ▶ Press **Reson** (State Variable Filter).
- ▶ Turn the **Encoder** to [ 75.0% ].
  - // The S.V. Filter is now working as a narrow band-pass, selecting a frequency band for the feedback loop.
- ▶ Press **Cutoff** (State Variable Filter).
- ▶ Sweep the **Encoder** slowly across the entire range and dial in a value that pleases your ear, let's say [ 80.0 st ].
  - // Shaping the feedback response using the S.V. Filter produces stunning timbral results. By shifting the bandpass, self-oscillation appears only when the band is matching one of the overtones that the Comb Filter can

produce. Sweeping the S.V. Filter Cutoff will generate a pattern of overtones. Please keep in mind that all you are hearing is the output signal of the Comb Filter – the S.V. Filter is just part of the feedback path (between Comb Filter and Feedback Mixer) and provides a selective feedback signal. Oscillator A excites the Comb Filter and is not audible as such either.

### ... by using the effects output as feedback signal:

Another interesting way to shape comb filter / physical modeling sounds of the C15 is using the feedback path of the effects section.

First, disable the S.V. Filter in the feedback path of the Comb Filter (of course, the Feedback Mixer provides several feedback paths in parallel but, for the time being, we want to keep things simple):

- ▶ Press **S.V. Filter** (Feedback Mixer).
- ▶ Turn the **Encoder** to [ 0.0% ].

### Feeding back signals from the Effects section to the Comb Filter:

- ▶ Press **Effects** (Feedback Mixer).
- ▶ Turn the **Encoder** slowly up and dial in a value that generates a mild feedback sound. Values around [ 50.0% ] should work fine.
- ▶ Press the **Mix** parameter of every effect and dial in a high mix value.
  - // Now you are hearing the feedback signal of the effects chain exciting the comb filter. While doing so, you will (hopefully) be surprised by some staggering soundscapes. Each of the effects individually provides a different treatment of the feedback signal and thus contributes a different result to the audible sound. Cabinet may be used to alter the harmonic content while the Gap Filter (which is a band reject filter that cuts out a certain frequency range) is useful to control the frequency response of the feedback signal. Flanger, Echo, and Reverb generally add different spatial components and motion to the sound. Please note that the amount of reverb in the feedback path can be adjusted separately by the **Rev Mix** parameter of the Feedback Mixer.

### ✓ Recap: The Feedback Paths

- Together with the Oscillator / Shaper sections and the Comb Filter, the feedback paths of the C15 provide interesting physical modelling capabilities.
- Using feedback paths produces sustained tones without using sustaining oscillator (exciter) settings – great for sounds with woodwind, brass, and bowed-strings-like character.
- To set up a feedback path, select and enable a source signal within the Feedback Mixer and a FB Mix point in the Shaper sections. The polarity of the feedback amounts can be crucial to the sound.
- The Drive parameters of the Feedback Mixer can shape the feedback sound.
- Altering exciter settings (Oscillator A and its Envelope A) also has an influence on the resulting sound.
- The State Variable Filter can be used to select overtones for self-oscillation.
- The output signals of the effects can also be fed back via the Feedback Mixer.