MANUALONE

A Single, Comprehensive Guide to Frap Tools' Modules

I'd like to thank a few precious friends for their support during these years, for the discussions we had, and for the valuable feedback i received — in alphabetical order: Tina Aspiala, Caterina Barbieri, Sebastian Baumann, Alessandro Bonino, Marco Ciccotti, Enrico Cosimi, Lorenzo Florissi, Tom Hall, René Margraff, Gianfranco Marongiu, Chris Meyer, Giulio Saltini, Alessio Santini, Stephan Schmitt, Brian Smith, Trevor Tunnacliffe, Giona Vinti, Andreas Zhukovsky.

And thanks to the amazing people I'm working with — in alphabetical order: Fabrizio Benatti, Federico Foglia, Giovanni Grandi, Antonio Masiero.

Simone Fabbri

TABLE OF CONTENTS

TABLE OF CONTENTS **3**

SAFETY AND WARRANTY **6**

BEFORE STARTING **7**
1 Connecting the Power 7
2 Mounting the Module 7
3 Warm-Up and Working Temperature 7

MODULAR SYNTHESIS: CORE CONCEPTS **8**
1 Voltage (and Current) 8
2 DC And AC 8
3 Audio and CV 8
4 Timing Pulses 9
5 Polarity 9
6 Audio and CV Processing 9

SUGGESTED READINGS **10**

INTERFACES **11**
1 Arrows (Input, Output) 11
2 Square and Round Shapes 11
3 Lines (Solid, Dotted, Dashed) 11
4 Color Coding 12
5 Combinations 12

CGM – CREATIVE MIXER SERIES **13**
1 Philosophy and Design 13
2 System Setup (Linking) 14
2.1 Link System 14
2.2 Master to Group(s) 14
2.3 Group to Channel(s) 15
3 Channel 15
3.1 Amplitude Controls and Direct Output 15
3.2 Effect Sends 16
3.3 Pan 16
3.4 Creative Functions 16
3.4.1 Mute 16
3.4.2 Solo in Place 16
3.4.3 PFL 16
4 Quad Stereo Channel 16
4.1 Amplitude Controls and Direct Output 16
4.1.1 Mono Auxiliary Input 17
4.2 Effect Sends 17
4.3 Pan and Crossfade 17
4.3.1 Pan 17
4.3.2 Crossfade 17
4.4 Creative Functions 17
4.4.1 Mute 18
4.4.2 Solo in Place 18
4.4.3 PFL 18
5 Group 18
5.1 Amplitude Control 18
5.2 Effect Sends/Returns 18
5.2.1 Sends 18
5.2.2 Returns 18
5.3 Group Output 18
5.3.1 Group Jumpers Configurations 18
5.4 Creative Functions 19
5.4.1 Group Mute 19
5.4.2 Safe Solo 19
5.4.3 PFL 19
6 Master 19
6.1 Amplitude Control 19
6.2 Auxiliary Stereo Input 19
6.3 Creative Functions 19
6.3.1 Headphones 19
6.3.2 PFL 19
7 Technical Data 20
7.1 Flow Charts 20
7.2 Specifications 22

SAPÈL **23**
1 Philosophy and Design 25
2 Noise Outputs 25
3 Voltage Sampling 25
3.1 Internal Clock and Clock Modulation 25
3.2 External Clock 26
3.3 Clock Mix 26
3.4 Manual Sampling 26
3.5 External Gate Sampling 26
3.6 Clock Outputs (Main and Random) 26
4 Random Voltages 26
4.1 Non-Quantized Random Voltages 27
4.2 Quantized Random Voltages 27
4.3 Fluctuating Random Output and Global Rate of Change (random clock density control) 28
5 Probability Distribution (Stored Random Voltages) 28
6 Technical Data 29
6.1 Flow Chart 29
6.2 Specifications 29

FUMANA **30**
1 Philosophy and Design 31
1.1 Spectral Transfer: A Brief History 31
1.2 Panel Overview 31
2 Audio Inputs 32
3 Audio Outputs 32
4 Audio Processing and Modulation Path 32
4.1 Faders and CV 33
4.2 Macro Spectral Editing 33
4.2.1 Tilt 33
4.2.2 Parametric Scanning 33
4.3 Spectral Transferring: Modulation Filters and Envelope Followers 33
4.4 The 'Unvoiced' Section 34
5 Filter Design 35
6 Patch Examples 35
6.1 16-Band Spectral Transfer 35
6.2 Dual 8-Band Spectral Transfer 35
6.3 Hybrid Spectral Transfer 35
6.4 Vocoder-Like Behavior 35
7 Technical Data 37
7.1 Transfer Function 37
7.2 Specifications 38

FALISTRI **39**
1 Philosophy and Design 39
2 Function Generators 39
2.1 Times 40
2.2 Shapes 40
2.3 Trig and Modes 40
2.3.1 Green Alternative Retrig (On Rest) 41
2.4 Outputs 41
2.5 Additional Generator Features 41
2.5.1 Quadrature 41
2.5.2 Max 42
2.5.2.1 ADSR 42
3 Function Processors 42
3.1 Dual Cascaded Frequency Divider 42
3.2 Four-Quadrant Multiplier 43

3.2.1 Amplitude Modulation & Ring Modulation (2 vs 4 quadrants) 43
3.2.2 Trimming 44
3.3 Linear Slew Limiter 44
4 Technical Data 45
4.1 Specifications 45
4.2 Revisions 45

USTA 46
1 Quick start 46
2 Philosophy And Design 47
2.1 Architecture 47
2.2 Tempo Management 47
3 Basic Editing and Visual Feedback 48
3.1 Editing Projects – Project Menu 48
3.2 Editing Tracks – Track Menu 49
3.2.1 Clock Settings 49
3.3 Editing, Playing and Looping Patterns 49
3.4 Editing Stages 50
3.4.1 Length 51
3.4.2 Maintain Pattern Length on Variation 51
3.4.3 CV Layers 51
3.4.3.1 Red CV Layer: Value 52
3.4.3.2 Green CV Layer: Variation Index 52
3.4.3.3 Blue CV Layer: Variation Range 53
3.4.4 CV Stage Colors 53
3.4.4.1 USTA Slide vs FALISTRI Slew 53
3.4.5 Gate Layers 54
3.4.5.1 Red Gate Layer: Value 54
3.4.5.2 Green Gate Layer: Variation Index 54
3.4.5.3 Blue Gate Layer – Variation Range 54
3.4.6 Gate Stage Colors 54
4 Quick Editing 55
4.1 Set All and Shift All 55
4.2 Coarse and Fine 56
4.3 Combining the Buttons 56
5 Performing 56
5.1 Play/Pause, Reset and Master Track Settings 56
5.1.1 Master Track 56
5.1.2 Play/Pause 56
5.1.3 Reset 57
5.2 Stage Loop 57
5.2.1 Infinite Stage Loop 58
5.3 Song Mode 58
5.3.1 Pattern to song while playing (and vice versa) 59
5.4 Live Performance Tools: Pattern Recall 59
5.4.1 Full pattern recall 59
5.4.2 Pattern Mix 60
5.5 Mute 60
5.5.1 Mute Track 60
5.5.2 Mute Channel 60
5.6 Hold 60
6 Advanced Editing 61
6.1 Composition Mode 61
6.2 Use an External CV/Gate Keyboard 61
6.3 Store Pattern: Last Played or Last Full 61
6.4 Rotate Pattern 62
6.5 Cloning 62
6.5.1 Clone a Stage 62
6.5.2 Clone a Structure 63
6.5.2.1 Clone a Layer 63
6.5.2.2 Layer Cross-Cloning 63
6.5.2.3 Clone a Pattern 63
6.5.2.4 Clone a Track 63
6.6 Quick Track Initialization 64
6.7 Quick Song Initialization 64
7 External Controls 64
7.1 Clock Input 64
7.2 Auxiliary Trig/Gate Input 64
7.2.1 Reset 64
7.2.2 Run 64
7.3 External CV 64
7.3.1 Pitch Shift 65
7.3.2 Root Shift 65
7.3.3 Gate Shift 65
7.3.4 Stage Shift 65
7.3.5 Vari Shift (Variation Shift) 65
7.3.6 Pattern Shift 66
7.3.7 Phase Shift 66
8 Additional Operations 66
8.1 Select CV Mode (Raw or Pitch) 66
8.2 CV Range 67
8.3 Gate Width % 67
8.4 Swing 67
8.5 Current Stage Data 67
9 Playing in Tune 68
9.1 Root & Scale – Dynamic Quantization 68
9.2 Microtonalities 68
9.3 Custom Scales 69
9.4 Set the Reference Note 70
9.5 Custom Temperaments 70
9.5.1 Absolute or Relative Temperaments 70
9.6 LED Pitch Tables 71
10 Additional Maintenance 73
10.1 Remove / Insert the SD Card 73
10.1.1 Remove the SD Card 73
10.1.2 Insert the SD Card 73
10.2 Analog Trimming 73
10.3 Project Management and Backups 73
10.4 Firmware Update 73
11 CHARTS 74
11.1 Project Menu 74
11.2 Track Menu 74
11.3 Project Hierarchy 75
12 Scale Tables 76
13 Change Log 83
14 Technical Data 84
14.1 Specifications 84

BRENSO 85
1 Philosophy, Design and Signal Flow 86
2 Frequency 86
2.1 Oscillators 86
2.1.1 Fine and Coarse Tuning 86
2.1.2 Coarse Frequency Lock 86
2.1.3 V/oct and Integrator 87
2.2 Frequency Modulation 87
2.2.1 FM Routing 88
2.3 Sync 88
2.3.1 Lock 89
2.3.2 Flip Sync 89
3 Timbre 89
3.1 Triangle Shaper 89
3.2 Pulse Shaper 90
3.2.1 Pulse-Width Modulation (PWM) 90
3.2.2 Waveshaper 90
3.3 Wavefolder 90
3.3.1 Sources 91
3.3.2 Folding 91
3.3.3 Symmetry 91
3.3.4 Ping 91
3.4 Timbre Modulation Bus 92
4 Amplitude 92
4.1 Amplitude Modulation and Ring Modulation 92
5 Trimmers 93
5.1 Accessible Trimmers 93

5.1.1 Coarse Frequency 93
5.1.2 Sine Wave Symmetry 93
5.1.3 Sawtooth Wave Symmetry 93
5.1.4 Exponential FM Zero 93
5.1.5 Triangle Waveshaper Shape 93
5.1.6 Wavefolder Symmetry 93
5.1.7 Four-Quadrant Multiplier 93
5.1.8 Comparator 93
5.2 Non-Accessible Trimmers 93
5.2.1 Gain 94
5.2.2 Base 94
5.2.3 Symmetry 94
6 Technical Data 94
6.1 Simple Signal Flow 94
6.2 Specifications 95
6.3 Revisions 95

333 96
1 Design 96
2 Technical Data 97
2.1 Flow Chart 97
2.2 Specifications 97

321 98
1 Design 98
2 Technical Data 99
2.1.1 Flow Chart 99
2.1.2 Specifications 99

WHAT'S IN THE BOX 100

LIST OF REVISIONS 101

SAFETY AND WARRANTY

The Frap Tools srls warranty covers the following products (hereinafter 'Frap Tools'), for two (2) years following the date of purchase. This warranty covers any defect in the manufacturing of this product. This warranty does not cover any damage or malfunction caused by incorrect use as described in the following instructions.

The warranty covers replacement or repair, as decided by Frap Tools. Please contact customer service at hello@frap.tools for a return authorization.

Frap Tools warrants that your new Frap Tools product, when purchased from an authorized Frap Tools dealer, shall be free of defects in materials and craft for a period of two (2) years from the original date of purchase. Please contact Frap Tools for warranty and service outside of Europe. During the warranty period, Frap Tools shall, at its sole option, either repair or replace any product that proves to be defective upon inspection by Frap Tools. Frap Tools reserves the right to update any unit returned for repair and to change or improve the design of the product at any time without notice. This warranty can be transferred to anyone who may subsequently purchase the product provided that such transfer is made within the applicable warranty period and that Frap Tools is provided with all of the following items:

· all warranty registration information for the new owner;

· proof of the transfer within thirty (30) days of the transfer purchase, and a photocopy of the original sales receipt.

Frap Tools shall determine warranty coverage in its sole discretion: this is your exclusive warranty. Service and repair of Frap Tools products are to be performed only by Frap Tools or an authorized service company. Unauthorized service, repair, or modification will void this warranty.

Please follow the given instructions for the use of the device because this will guarantee the correct device operation. Since these instructions also include indications concerning Product Liability, they must be read carefully. Any claim for defect will be rejected if one or more of the following points is not observed. Any disregard of these instructions can void the warranty.

The devices may only be used for the purpose described in this operating manual. Due to safety reasons, the devices must never be used for purposes not described in this manual. If you are not sure about the intended purpose of the devices, please contact an expert or Frap Tools at the email address above.

Do not use or store the devices in humid places. Avoid contact with any liquid.

Do not touch any component of the devices when it is power or connected to any power source.

Do not place the devices on unstable carts, stands, tripods, tables, or other surfaces, or on surfaces that are not perfectly plane. Such behavior may cause the devices to fall, which could result in human injury, property damage or improper functioning of the devices themselves.

The devices are designed for use only when safely and tightly mounted in a proper Eurorack case, made of non-flammable materials. If you are not sure about the intended purpose of the devices, please contact an expert or Frap Tools at the email address above.

Do not ever leave the devices switched on when not in use.

To prevent fire, never place any candle, flame, or other sources of heat on or near the devices.

Transport the devices only in the original box with original packaging or when safely and tightly mounted in a proper Eurorack case and handled with care. Never let the devices fall or topple. Make sure that during transport and while in use the devices and their case, have a proper stand and do not fall, slip or turn over because of potential human injury to persons or property damage. Any damage from physical abuse such as dropping the unit, impact from hard objects or damage to external components as a result of negligence will void this warranty.

Never expose the devices to temperatures above +40°C or below 0°C.

Before any operation, also verify the operating temperature ranges of all the modules and the power boards in use. Do not keep or leave the case that hosts the device, or the devices themselves near heat sources.

Any modification must be carried out only by Frap Tools or an authorized service company. The devices may not be modified in any way by any parties not expressly authorized by Frap Tools. Any repair, modification, tampering, or attempted repair made by unauthorized personnel will void this warranty.

Frap Tools cannot be held responsible in any way for problems to persons or property or to the devices themselves, if the devices are installed improperly, or if they are improperly used, maintained, or stored.

Any device shipped to Frap Tools for return, exchange, warranty repair, update, or examination must be sent in its original packaging! Any other deliveries will be rejected. Therefore, you should keep the original packaging, and any technical documentation or manual provided. The device must be shipped only with the original packaging. As specified on the product box, this box is not intended for shipment: if you bought the device directly at a physical reseller's shop, you should put the device in the original packaging and put the packaging in a properly larger box with proper packaging destined for shipping. If you received the device via carrier or any post service, it should have come with a proper double box packaging.

All non-warranty services are subject to a minimum fee of €50.00+VAT (within the European Union). The customer must pay for shipping to Frap Tools; Frap Tools will cover return shipping costs.

It is important to note that the front panel of our modules may get warm and may warm up the case where it is mounted. Please do not be alarmed, as this is normal and is part of its standard operation.

Shut down your equipment immediately if it produces smoke, a strange odor, or unusual noise. Continued use may lead to fire. Immediately unplug the equipment and contact your dealer or Frap Tools at the address above for advice.

Never attempt to repair this product yourself. Improper repair work can be dangerous. Never disassemble or modify this product. Tampering with this product may result in injury or fire and will void your warranty.

Do not allow foreign matter to fall into the equipment. Penetration by foreign objects may lead to fire.

If water or other liquid spills into this equipment, do not continue to use it. Continued use may lead to fire. Unplug the power cord immediately and contact your dealer or Frap Tools at the address above for advice.

The internal components of our modules and power supplies can get very hot. Do not touch any internal components while it is connected and/or powered and after they completely cool down after use for at least 30 minutes.

BEFORE STARTING

1 CONNECTING THE POWER

To connect the power cable, carefully follow these two rules:

· the power connector on the module is the keyed one in the top;

· the red line on the cable should be placed matching the −12V side on your power board: please double check with your power boar supplier that the marked side is the −12V.

Frap Tools may not be held responsible in any way for problems or damage to persons or property or to the device itself, if the device is not connected as indicated above.

2 MOUNTING THE MODULE

After connecting the power as explained in the previous section, install the module in your case using all the 2 or 4 screws provided. Make sure that the module is safely and tightly connected to your Eurorack case.

Frap Tools modules use the standard Eurorack orientation and color-coding: the red line on the power cables is placed at the bottom and stands for the −12V. Please double check with the power system you want to use that it adopts the same powering system.

Frap Tools may not be held responsible in any way for problems or damage to persons or property or to the device itself, if the device is not connected as indicated above.

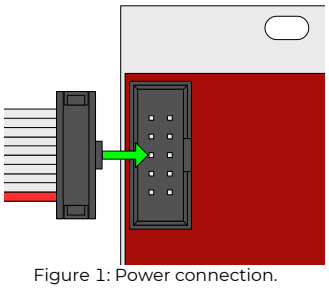


Figure 1: Power connection.

3 WARM-UP AND WORKING TEMPERATURE

For best performances, we suggest letting the Frap Tools modules warm up at least around 20 minutes prior to use it [tested at 25°C]. It is absolutely normal that they feel warm when touched.

In this introduction we'll go through the basics of analog modular synthesizers in their broader meaning, to make sure that all the technical jargon that we'll be inevitably using throughout the manual will be, hopefully, clarified for the newcomers.

Modules in a modular synthesizer can be thought of as strangers sitting next to each other on a train compartment, sitting in an embarrassed silence.

Creating music from a modular synthesizer is like starting a conversation on our hypothetical train compartment, by connecting modules with patch cables and making them "speak" to each other.

For any conversation to happen we need two things: the people must have something to say (like an argument), and they must speak the same language.

In modular synthesizers, the argument is your musical idea, and it is beyond the scope of this manual, while the common language to make your modules communicate is the *voltage*.

1 VOLTAGE (AND CURRENT)

Even though this is not an engineering manual, a general introduction on the concept of voltage might be useful to better understand how these instruments work.

Voltage is the difference in electric potential between two points of a circuit, or, oversimplifying, «what you apply to cause currents to flow.»[1] It is measured in volts, whose symbol is V. Without any other specification, the voltage is calculated between any point of the circuit and a reference point called *ground* (0V).

Electric potential brings in the concept of *current*, which is the actual flow of electric charge through a circuit. Electric charge is measured in coulomb (C), while the current is measured in amperes (A): 1 ampere equals to 1 coulomb of charge passing through a given point in 1 second.

Enough for the technical stuff: now it's time to see how voltage and current turn into (electronic) music.

2 DC AND AC

There are two kinds of current, each of which implies a different kind of voltage. They are *direct* and *alternating* current, abbreviated *DC* and *AC* respectively, and we use both in modular synthesizers.

In case of direct current, our charge flows only in one direction: if the current is steady, so will do the voltage. This kind of fixed voltages are what we use, for example, to send note information to the sound source.

In case of alternate current, our charge flows back and forth in our circuit. Its alternation over time can be displayed with a waveform diagram. In case of alternate current, the voltage changes over time as well. We use this kind of varying voltages to generate, for example, waveforms that ultimately will turn into sound.

3 AUDIO AND CV

To hear music, we need some vibrating air. To make the air vibrate, we need a vibrating body: we can pluck a string, hit two wood sticks, blow into a pipe and so on.

In case of electronic instruments, our vibrating body is the loudspeaker (or the headphones).

More in detail, in the specific case of analog modular synthesizers, everything, from the sound generation, to its articulation, to the final audio output through the speaker cone' vibration, is achieved through voltage that change over time.

With our modular synthesizer we create a complex, electric signal that makes our cones move back and forth. The variation of voltage over time translates into the complex waveform that we call music.

It goes without saying that the more control we have over our voltage, the more expressive and articulated will become our composition's waveform.

In modular synthesizers, we use voltage to generate sound, but also to control it.

If a circuit generates a voltage that varies from, say, -5V to 5V, 110 times a second, it will make our loudspeaker vibrate as many times, thus producing a sound whose *frequency* is 110 Hertz (Hz), corresponding to the note A.

However, if the same circuit generates a voltage oscillation of 10Hz, meaning that the voltage varies from -5 to 5V ten times a second, we wouldn't be able to hear it anymore. It's because its frequency is below our audible range of human beings, that spans from 20 to 20.000Hz.

This brings in a conventional distinction between voltages that we use for sound *generation*, which provide alternate current in the audio range, and voltages that we use for sound *modulation* or *control*, which are often called *control voltages*, or CV. Generally speaking, we use CV to turn our modules' knobs for us, in a sort of automated way.

CVs can be of any kind: they can generate alternate current, such as low-frequency oscillators (LFOs, like the ones we use for tremolo effects) or direct current (like the fixed CVs that we use to send pitch information).

It is nice to point out that in modular synthesis we use the same two or three principles to take care of every *parameter* of the sound design.[2]

[1] Horowitz 2019, 1, note 2.

[2] On the importance of *parametrical* thinking when approaching analog sound design, see especially Strange 1984, 4–5.

4 TIMING PULSES

We use CV to distribute information (like pitch, amplitude, timbre) all over our electronic composition.

For example, the information of "C#4" is not enough to generate a note: it generates just a pitch. To have a note, we need to give it some timing: when it begins, how long it lasts, when it ends.

For this kind of information, which is essential to define any kind of musical event (note, melody, rhythm), we use a particular kind of CV called *timing pulses*, which are *trigs* and *gates*.[3]

Trigs and gates are *pulses*, because they can have only two values, off and on, and because the transition between one value and the other is almost immediate. *Off* is usually 0V, *on* is around 5V.

The main difference between trigs and gates is that trigs are very fast voltage bursts that go from 0 to 5V and immediately (actually, after one or two milliseconds) back to zero, while gates can stay at 5V as much as we want.

A trig is thus a timing pulse that defines when a certain musical event should happen, while a gate is a timing pulses that tells when such an event should happen, but also exactly for how long.

Trigs are the kind of pulses that we get out of clock generators, such as SAPEL, while gates are generated by modules such as FALISTRI or USTA, which are more focused on articulation and dynamics.

5 POLARITY

We have said that voltage is a difference in electric potential that causes the current to flow through a circuit. Such a current can flow through a point in either direction, depending on the voltage that we apply to the circuit. A positive voltage will make the current flow in a direction, and a negative one in the opposite. (0V will not have any current flow, and such is the value of the circuit's *ground*.)

In case of alternating current, the voltages are also alternating. If such alternation oscillates above and below 0V, the voltage would be *bipolar*. However, there are cases in which the change in voltage is only positive (or, more rarely, negative). In such cases, we say that the voltage is *unipolar*.

The most common bipolar voltages are the ones used for generating audio waveforms, since they need to move the speaker's cone back and forth. A unipolar audio signal would make the cone move in one direction only, with potential damages.

Unipolar voltages are, for example, certain envelopes, or LFOS, especially the ones used for controlling a parameter which needs to operate in one direction only, such as amplifiers (on which see the next chapter).

[3] Strange 1984, 51–52, 61–62.

6 AUDIO AND CV PROCESSING

In modular synthesis, we add dynamics and expressiveness to our music through signal processing. We use quite a few concepts, but their combination provides many different results.

The most important element of signal processing is the control over a signal's *amplitude*. Amplitude is, roughly, how loud we perceive the sound, which means how much the air vibrates, which also means how much our speaker cones vibrate, which, in turn, means how high is the voltage oscillation that we generate with out synthesizer.

Controlling the amplitude of a signal means being able to define the voltage range of our sound source, which is bipolar: for example, an amplitude of 5V means that our waveform will go from 0V to 5V, and then from 0V to -5v throughout each cycle. Since there is a difference of 10V from 5V to -5V, we can also say that our signal has an amplitude of 10V peak-to-peak, i.e., measured from the highest to the lowest point of the waveform cycle ("peaks").

The circuit that allows us to control the amplitude is called amplifier: increasing or decreasing the amplifier value, usually through a knob, increases or decreases the amplitude of our sound, and eventually brings it to 0V, where no sound is passing through the circuit. An amplifier is often capable of increasing a signal's amplitude, thus providing a louder sound than the one generated by our sound source. A circuit that only reduces a signal's amplitude is often called attenuator.

Furthermore, amplifiers can also deal with signals that are not waveforms, such as control voltages. In this case, amplifiers control the voltage magnitude, and they can make controls sources and their modulations more or less effective in a patch.

An amplifier that is we can control through voltage is called VCA, voltage-controlled amplifier. For example, if we control an amplifier with an LFO, this will cyclically increase and decrease the signal's amplitude, like a tremolo effect. But we can also control the LFO's amplitude through a second VCA, and now it will change the modulation depth.

Another useful concept to outline now is *inversion*, which relies on the fact that voltages, in Eurorack synthesizers, can be positive or negative. Inverters are circuits that change the polarity of a given signal: on a graphical representation, the result of an inverted continuous voltage will be a mirrored, upside-down version of the original signal.

If the signal is bipolar, the positive values will become negative, and vice-versa.

We can also invert audio signals, and in such case we'll invert their *phase*

SUGGESTED READINGS

Bernstein, David W. *The San Francisco Tape Music Center. 1960s Counterculture and the Avant-Garde*. Berkeley: University of California Press, 2008.

Bjørn, Kim. *Patch & Tweak. Exploring Modular Synthesis*. Copenhagen: Bjooks, 2018.

Bode, Harald. "History of Electronic Sound Modification." *AES: Journal of the Audio Engineering Society* 32, no. 10 (1984).

Chowning, John M. "The Synthesis of Complex Audio Spectra by Means of Frequency Modulation." *AES: Journal of the Audio Engineering Society 21*, no. 7 (1973): 526–34.

https://web.eecs.umich.edu/~fessler/course/100/misc/chowning-73-tso.pdf

Hordijk, Rob. "Designing Instruments for Electronic Music." *EContact!* 17, no. 4 (2016).

https://econtact.ca/17_4/hordijk_design.html

Horowitz, Paul. *The Art of Electronics*. Third edition. New York: Cambridge University Press, 2019.

Roads, Curtis. *The Computer Music Tutorial*. Cambridge, MA-London, 1996.

Shapiro, Peter. *Modulations. A History of Electronic Music. Throbbing Words on Sound.* New York: Caipirinha Productions, 2000.

Strange, Allen. *Electronic Music: Systems, Technics, and Controls.* Second edition. Dubuque: Brown, 1984.

Wells, Thomas. *The Technique of Electronic Music.* Second edition. New York-London: Schirmer Books-Collier MacMillan, 1981.

INTERFACES

Here at Frap Tools, we put a lot of effort into designing a proper user interface for each of our modules.

By "proper user interface" we mean essentially three things:

1. it must convey the module's identity at a glance;
2. it must allow for a smooth creative workflow;
3. it must be pleasant to look at.

Moreover, we want our interfaces to be clear, but not self-explanatory (or, in other terms, cryptic, but not chaotic).

The reason for doing so is that, in our vision, the musician should master the "code" of the instrument before playing it: a piano does not have the note names on its keys, a violin does not have marks on its neck — it is up to the musician to practice and learn how the instrument works.

In the same way, our modules do not have labels such as 'frequency' or 'decay time': instead, they are replaced with a system of symbols and colors that try to be as consistent as possible. Moreover, a musician approaching our modules through a "conventional" labeling system might be tempted to assume that the module behaves in an ordinary way, which sometimes is not completely correct.

The modules are explained in detail further in this manual. However, to allow the musician to get acquainted with the overall symbol system, we provide here a brief guide to "decode" the most recurrent elements of the "Frap environment".

1 ARROWS (INPUT, OUTPUT)

An arrow can mean either an input or an output, according to its position: if it points towards one or more jack sockets, it is an input; if it points away from one or more jack sockets, it is an output.

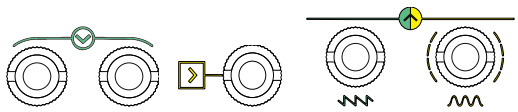


Figure 2: Arrows.

2 SQUARE AND ROUND SHAPES

All the modular world revolves around voltage. The most basic distinction is, conventionally, between voltage used for timing pulses (trigs or gates) and voltage used for audio signals or CV.

The former is a discrete signal with two levels only ("off" and "on", "low" and "high"). In our modules it is associated with square shapes: the reason is that squares have only two kind of lines (vertical and horizontal), in the same way a gate or rig signal has only two possible states, on and off.

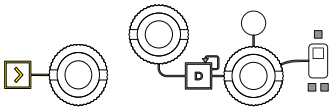


Figure 3: Square shapes.

The latter is a continuous signal (or 'analog' in its closest meaning). Is associated with round shapes because circles can be thought of as having infinite sides, in the same way an analog signal has infinite values.

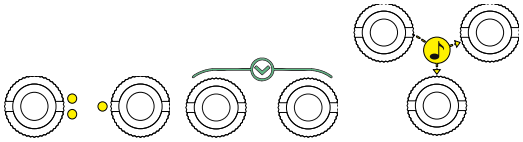


Figure 4: Round shapes.

A subgroup of audio analog section is the stereo audio. As you can notice in the CGM series, the group and master modules feature stereo in and out out: here, the left/mono is connected to the solid-colored area, while the right is connected to the ring that surrounds it. The reason is that the left output is always the primary (because it can be used as a mono output as well), while the right one is more of an accessory to it.

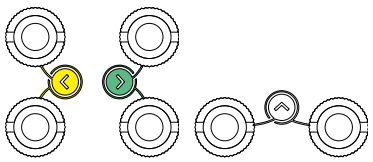


Figure 5: Stereo audio.

3 LINES (SOLID, DOTTED, DASHED)

A solid line relates two or more elements of the circuit. It stands for manual control, which means that a given knob or switch directly affects the signal passing through the circuit from an input or to an output.

Figure 6: Solid lines.

A dotted line stands for external CV control, and it often relates a jack socket to a manual control such as a knob or a slider. It means that the specific parameter can be voltage controlled.

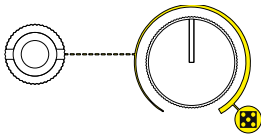


Figure 7: Dotted lines.

A dashed line relates two or more inputs or two more outputs: it means that they are semi-normalled, or, in other terms, that the signal going to one input or coming from one output is mirrored by the other jack sockets connected by a dashed line. Such behavior is automatically overridden once a cable is plugged to another jack socket (thus "breaking" the normalization).

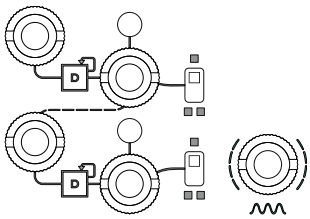


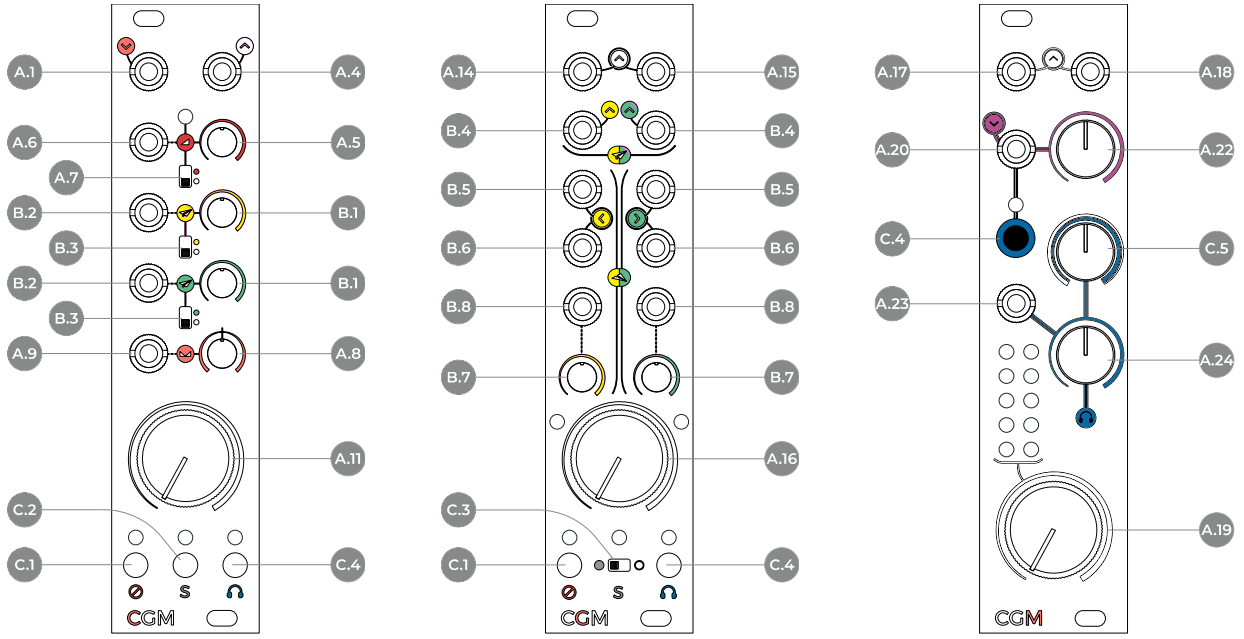
Figure 8: Dashed lines.

4 COLOR CODING

As a rule of thumb, if a module performs more than one function, the respective controls are marked with different colors. In other words, a given color relates to one and only one section of the module design. In case a module features two "mirrored" sections (such as SAPÈL's or FALISTRI's generators, or CGM Group's FX sends), they are marked in yellow and green.

5 COMBINATIONS

All the elements can be combined. For example, an arrow within a square pointing towards a jack socket means that it is a gate/trig input; if a dotted line connects a jack socket to a knob, and a solid line connects the same knob to another jack socket, it means that the signal outputted from the second jack socket can be modified either manually via the knob or automatically via an external CV patched to the first socket.



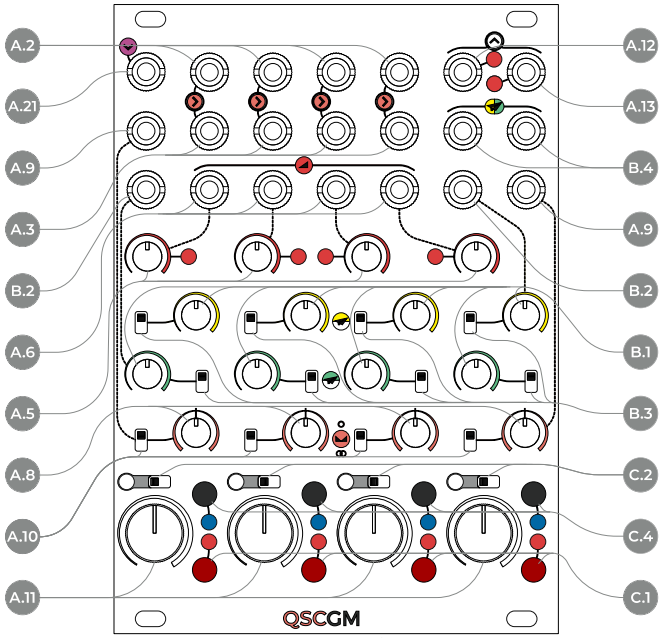


Figure 9: CGM Interface

A Main Audio Routing
A.1 Audio Mono Input
A.2 Audio Input Left/Dual Mono
A.3 Audio Input Right/Dual Mono
A.4 Direct Out
A.5 VCA Level
A.6 VCA Level CV Input
A.7 VCA Pre/Post
A.8 Pan Level
A.9 Pan Level CV
A.10 Pan/Crossfade Switch
A.11 Channel Fader
A.12 Local Output Left
A.13 Local Output Right
A.14 Group Output Left
A.15 Group Output Right
A.16 Group Fader
A.17 Master Output Left
A.18 Master Output Right
A.19 Master Fader
A.20 Aux. Stereo Input
A.21 Aux. Mono Input
A.22 Aux. Input Level
A.23 Phones Stereo Output
A.24 Phones Level
B Send/Return
B.1 Effect Send Level
B.2 Effect Send Level CV
B.3 Effect Send Pre/Post
B.4 Effect Send Mono Output
B.5 Effect Return Left Input/Dual Mono
B.6 Effect Return Right Input/Dual Mono
B.7 Effect Return Level
B.8 Effect Return Level CV
C Creative Functions
C.1 Mute Button
C.2 Solo-In-Place Button/Switch
C.3 Safe Solo Switch
C.4 Pre-Fader Listen (PFL) Button and LED
C.5 PFL Blend

1 PHILOSOPHY AND DESIGN

The CGM Creative Mixer is a modular mixing solution for Eurorack systems. It breaks down the main parts of a classic mixer console into three families of modules (the *Channel*, the *Group*, and the *Master*), without sacrificing the pro audio quality of the signal treatment.

The CGM is designed for a performance-oriented use in modular synthesizers: its modular structure allows for tailoring the size and kind of the mixer to the exact setup it must fit into, while the CV inputs over all the key parameters encourage heavy patching and modulations.

It is currently composed of four modules: the Channel, the Quad Stereo Channel, the Group, and the Master.

2 SYSTEM SETUP (LINKING)

All the Channel modules can work as stand-alone units, but they unleash their real potential when connected to the Group and Master modules.

In a CGM system, all the modules share audio (Channels and Group share even power) via 10-pole IDC ribbon cables[4].

You can connect up to eight Channel modules to a Group via IDC cables, forming a Group section, and up to four Group sections to a Master module. Note that the Group module does not have a power socket: we designed the CGM so that, in any configuration, you will need to power only the Master (if present) and one Channel per Group section.

If you accidentally connect more than one Group to the PSU, fear not: it's not going to damage your system! It's just useless and may not be optimal for ground.

2.1 LINK SYSTEM

The IDC cables that connect the modules of the CGM series form the Link System. There are currently two kinds of cables: Master to Group(s) and Group to Channel(s).

The Master to Group cable allows you to connect one or more groups to a master channel. As of today, there are four Master to Group cable configurations that enable you to link the Master to one, two, three, or four Groups, respectively. By default, the Master module is shipped with a one-group connector. The other cables can be purchased separately.

The Group to Channel(s) cables allow you to connect one or more channels to a group. You need two parallel IDC cables to connect the Channels to their Group. Currently, three purchase options allow you to connect to one Group two, four, or eight channels, respectively. Since there are two kinds of Channel modules (the mono one and the Quad Stereo Channel), the cables come folded to fit both sizes with different HP width. When folded, the space between every IDC connector can be arranged to obtain 6HP: this is the ideal configuration for connecting the classic mono Channel modules.

If you want to connect one or more Quad Stereo Channel modules (QSC), simply stretch the cable: the connector will now have the extra length necessary for the 18HP modules.

Before the introduction of the QSC, there were other cable options. They are now discontinued, but they are still fully compatibles with the new modules. The older one-Group-to-four-mono-Channels cable is still shipping with any Group unit.

2.2 MASTER TO GROUP(S)

Every Master comes with one Link cable to connect one Group to it. You can connect up to four Groups to the Master using a proper link cable with more than one plug.

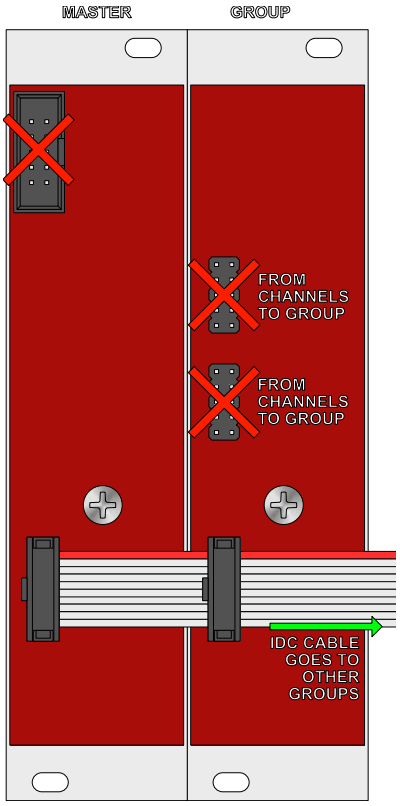


Figure 10: *Master* and *Groups* connection.

The Master module, on its back, has two 10-pole IDC connectors: the one at the top left corner is for power, while the other at the bottom are for Group linking. You can tell the difference between the two headers and the

[4] Refer to the *What's in the Box* section of the manual to see each module's content and have an overview of the available link cables for various setups, p. 90).

box: the power header is always boxed, while the Link headers are not.

To connect a master to a group, patch the link cable to both the Master and the Group's IDC connectors: they are placed at the same height to ease the connection.

In the example in

Figure 10, the cable continues and goes to other Groups.

2.3 Group to Channel(s)

Each Group comes with two link cables to connect one Group to up to four Channels, but you can connect up to eight Channels to a Group using a proper link cable.

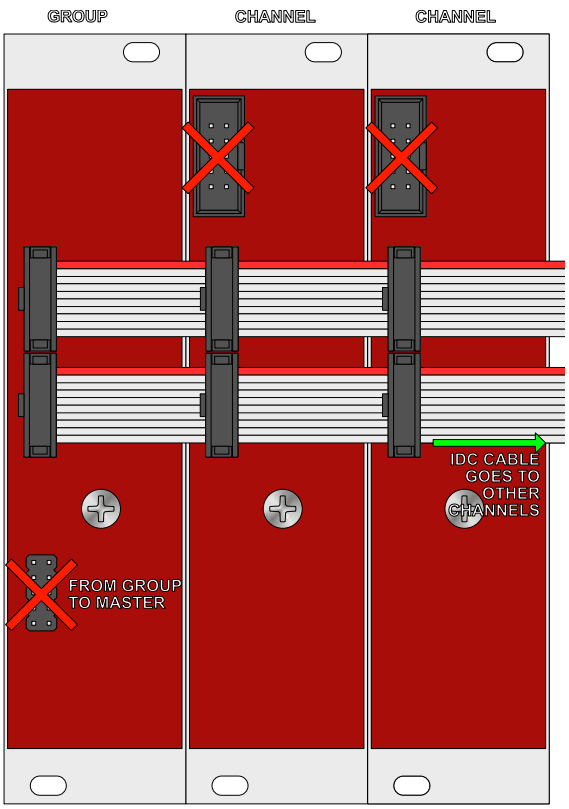


Figure 11: *Group* and *Channels* connection.

The main difference from the master-to-group connection is that the channels share with the group not just the audio but also the power: this is why at least one Channel is required for the Group to work.

Every Channel can be connected to one and only one Group. Connecting the same Channel to two Groups is not possible.

The Channel module, on its back, has three 10-pole IDC connectors: the one in the top left corner is for power, the other two in the center are for linking. You can tell the difference between the two headers and the box: the power header is always boxed, while the Link headers are not.

To connect your *Channels* to a *Group*, patch the two linking cables to every IDC connector on the back of the PCBs. Connectors are placed at the same height on all the modules to ease the connection. When properly mounted, you should see the two IDC cables perfectly parallel.

In the example in Figure 11, the cable continues and goes to other Channels.

It is imperative to do not swap connectors! Frap Tools may not be held responsible in any way for problems or damage to persons or property or to the device itself if the device is not connected as indicated above.

3 CHANNEL

The Channel module is the first member of the channel family. Its concept draws from the monophonic channel strip of a classic mixing console, with the crucial implementation of CV controls over every key parameter, which makes it fully suitable for the Eurorack environment.

The signal routing is designed to provide the best audio quality and can be broken down to five color-coded main parts: VCA (red), two effect sends (yellow and green), pan (pink), and fader (white).

3.1 Amplitude Controls and Direct Output

The main VCA defines the input gain of the signal patched to the channel input (A.1) through the *Level* knob (A.5). The *Channel Fader* (A.11) sets the volume of the audio signal to be sent to the Group module. In other words, the VCA controls the amplitude of the signal coming in, and the fader controls the amplitude of the signal going out.

These two controls together, just like in classic mixing consoles, allow you to define not only the amplitude of the signal in the mix but also its "color."

The input VCA goes ~6dB above unity gain, which means that it can add a very smooth saturation to the incoming sound, which is often translated to adjectives such as "fatter, punchier, warmer," and so on.

Furthermore, such input level can be voltage-controlled through the *VCA Level CV Input* (A.6), which accepts unipolar signals whose range can be either 0÷5V or 0÷10V. When a signal is patched to this input, the *Level* knob (A.5) becomes its attenuator.

Practice the *VCA Level CV Input* with this Technique:
Sidechain #1

The main VCA is equipped with a *Direct Output* (A.4), which allows you to take the incoming signal after it has been amplified (or colored) by the VCA. It is extremely useful in the case of multitrack recording or parallel processing of signals.

For this purpose, the *Direct Out* has a *Pre/Post Switch* (A.7), which lets you define if it should work pre-fader (upper position, red dot), or post-fader (lower position, white dot).

If set pre-fader, the signal coming out of the *Direct Output* will be straight after the main VCA, regardless of the *Channel Fader* position (A.11); if set post-fader, its amplitude will be determined by the channel fader as well, and will thus mirror the presence of the signal in the final mix.

Practice the use of the *Direct Output* with these Techniques:
FUMANA Feedback #1
FUMANA Feedback #2
FUMANA Feedback #3

3.2 Effect Sends

The two *Send Level* knobs (B.1) define the amplitude of the channel signal to be routed to the Group's effect sends mono outputs (B.4).

Just like the Main VCA, they can be voltage controlled by patching unipolar CVs (either 0÷5V or 0÷10V) to its CV input (B.2). This input allows you to automate the effects of each channel individually in a way that it is not possible on standard mixing consoles.

Both the effect sends are equipped with a *Pre/Post Fader* switch (B.3), which improves the module's usability. If set post-fader (lower position, white dot), signal sent to the external effects will be limited by the channel fader position (A.11). This option is useful when you want your dry/wet signal ratio always the same when you adjust a channel's volume.

If set pre-fader, the effect send will be defined by the knob or the CV only, thus being completely independent from the overall channel level. In this way, you can achieve some effect such as lowering the channel fader and leaving just the processed signal in the mix.

Another way of looking at the pre/post effect send is that the former is an absolute level, while de second one is relative (to the main fader, in this case).

3.3 Pan

The *Pan* control lets you distribute the processed signal across the stereo image through the Pan knob (A.8).

This parameter is provided with a CV input (A.9), which accepts both positive and negative signals (range −5V÷5V).

The more the positive CV increases, the more the signal is distributed to the right, and vice versa for negative CVs and the left channel.

Given that the Pan is a bipolar control, the CV input works slightly differently from the VCA and the Effect Sends. In this case, when a CV is patched to its input, the knob shifts the incoming CV towards the positive (right) or negative (left) side. In other words, it acts as an offset, not an attenuator.

3.4 Creative Functions

Every channel has three creative controls: the *Mute* button (C.1), the *Solo in Place* (C.2), and the Stereo PFL (Pre-Fader Listening, C.4).

3.4.1 Mute

The Mute button (C.1) closes the main VCA. It equals turning the VCA Level knob (A.5) all the way to zero. For this reason, muting a channel allows you to mute all of its outputs, including the effect sends.

3.4.2 Solo in Place

The *Solo in Place* button (C.2) selects the channels to keep active when the group is in *Safe Solo* mode (C.3, see below §5.4.2).

3.4.3 PFL

The *Stereo PFL* button (C.4) selects the channels sent to the *PFL* circuit on the *Master* module (C.5, see below §5.4.3).

4 QUAD STEREO CHANNEL

The Quad Stereo Channel (QSC) consists of four stereo channels, whit a design based on the Channel (§3) module that, however, also borrows some concepts from the Group (§0) and the Master (§6) modules. The color scheme reflects the same routing of the modules: VCA (red), two effect sends (yellow and green), pan (pink), auxiliary input (purple), and main faders (white).

The local sum outputs allow you to use the QSC also as a standalone stereo mixer with mono effect sends. The panning section can switch its behavior between stereo source panning and mono sources crossfading.

4.1 Amplitude Controls and Direct Output

The main VCAs defines the input gain of the signal patched to the channel inputs (A.2, A.3) through the *Level* knob (A.5). The *Channel Faders* (A.11) set the volume of the audio signal to be sent to the *Group* module. Just like in the Channel module, the VCA controls the amplitude of the signal coming in, and the fader controls the amplitude of the signal going out.

These two controls together, just like in classic mixing consoles, allow you to define not only the amplitude of the signal in the mix but also its "color."

The input VCA goes ~6dB above unity gain, which means that it can add a very smooth saturation to the incoming sound, which is often translated to adjectives such as "fatter, punchier, warmer," and so on.

Furthermore, such input level can be voltage-controlled through the *VCA Level CV Input* (A.6), which accepts unipolar signals whose range can be either 0÷5V or 0÷10V. When a signal is patched to this input, the *Level* knob (A.5) becomes its attenuator.

Practice the *VCA Level CV Input* with this Technique:
Sidechain #1

One of the differences between the QSC and the Channel module is that it features a local stereo output (A.12, A.13) instead of individual Direct outputs (A.4). This allows you to use the QSC as a stand-alone, four-channel stereo mixer.

4.1.1 Mono Auxiliary Input

The QSC also features an additional monophonic DC-coupled input (A.21). This is an audio input with no controls over the signal's amplitude, and its signal routing is very straightforward, adding the sound as is to the overall sum routed to the Group or the two local outputs (A.13, A.14).

The Mono Auxiliary input can be useful if you need a spare mono channel for an external source that does not need further amplitude processing.

4.2 Effect Sends

The two *Send Level* knobs (B.1) define the amplitude of the channel signal to be routed to the Group's effect sends mono outputs (B.4).

All the effect sends are equipped with a *Pre/Post Fader* switch (B.3), which improves the module's usability. If set post-fader (lower position, white dot), the amount of signal sent to the external effects will be limited by the channel fader position (A.11). This option is useful when you want your dry/wet signal ratio to be always the same when you adjust a channel's volume.

If set pre-fader, the effect send will be defined by the knob or the CV only, thus being completely independent from the overall channel level. In this way, you can achieve some effect such as lowering the channel fader and leaving just the processed signal in the mix.

Another way of looking at the pre/post effect send is that the former is an absolute level, while the second one is relative (to the main fader, in this case).

The green send of the leftmost channel and the yellow send of the rightmost channel can be voltage controlled by patching unipolar CVs (either 0÷5V or 0÷10V) to their CV input (B.2).

The QSC is equipped with two local effects mono outputs (B.4), one for the yellow and one for the green sends. They work like the group mono send, outputting the sum of the signals defined by the four channels' send knobs.

These outputs allow you to take advantage of the mono sends even if you have a smaller system without a group.

4.3 Pan and Crossfade

The *Pan/Crossfade* knob can perform two different tasks according to the Pan/Crossfade switch position.

4.3.1 Pan

When the Pan/Crossfade switch (A.10) is set to the lower position, the knob (A.8) acts as a stereo panorama control, defining the balance of the (mono or stereo) audio input across the left and right channels.

Two CV inputs (A.9) allows for external control of the leftmost and rightmost channels' Pan with positive and negative voltages (range −5V÷5V). The more the positive CV increases, the more the signal is distributed to right, and vice versa for negative CVs and the left channel.

Given that the Pan is a bipolar control, the CV input works slightly different than the ones of the VCA and the Effect Sends. In this case, when a CV is patched to its input, the knob shifts the incoming CV towards the positive (right) or negative (left) side. In other words, it acts as an offset, not an attenuator.

4.3.2 Crossfade

When the Pan/Crossfade switch (A.10) is set to the higher position, the knob works as a "dual mono" crossfade that blends the signals patched to the two inputs and outputs the same result across the left and right channel.

It can be useful, for example, for blending the sound of two waveform outputs of the BRENSO oscillator (see p. 86) for a punchier, mono bassline or to mix two elements of a drum pattern that do not need to be panned across the stereo image.

Setting all the stereo channels to the Crossfade can even let you use the QSC as an almost-eight-channel-mono mixer.

4.4 Creative Functions

Every channel has three creative controls: the *Mute* button (C.1), the *Solo in Place* (C.2), and the Stereo PFL (Pre-Fader Listening, C.4).

4.4.1 Mute

The Mute button (C.1) closes the main VCA. It equals turning the VCA Level knob (A.5) all the way to zero. For this reason, muting a channel allows you to mute all of its outputs, including the effect sends.

4.4.2 Solo in Place

The *Solo in Place* button (C.2) selects the channels to keep active when the group is in *Safe Solo* mode (C.3, see below §5.4.2).

4.4.3 PFL

The *Stereo PFL* button (C.4) selects the channels sent to the *PFL* circuit on the *Master* module (C.5, see below §5.4.3).

5 GROUP

The Group is a stereo module that blends the signals coming from the channels linked to it (up to eight) and takes care of their signal routing.

The group "puts into practice" all the information defined by the channel's controls: volume, effect send volume, stereo panning.

It consists of three sections: the amplitude controls (white), and the two effect sends and returns (yellow and green).

As of today, the Group is the only module that does not require power, as a Channel powers it through its IDC cable.

5.1 Amplitude Control

The main control is the *Group Fader* (A.16), which sets the Group's final volume. This is the overall amplitude of the sum of the connected channels and the returns.

It features a pair of peak LEDs on the top left and right of the knob that notifies when the left or right Channel (respectively) are clipping.

5.2 Effect Sends/Returns

This section of the module is the CGM interface for external signal processing. It consists of two specular sections, marked yellow and green, each of which features a mono output (Send) and a stereo input (Return).

5.2.1 Sends

The *Send* jack socket (B.4) outputs a mono signal, which is the sum of all the signals processed for that send by each Channel connected to the Group.

5.2.2 Returns

The *Return* section is divided into two specular columns, the left one for the yellow return, the right one for the green return.

The first two rows from the top are the *Return* inputs (B.5, B.6). They are semi-normalled, meaning that they can work either in stereo (when two cables are plugged) or in dual mono (when only one cable is plugged into any of the two inputs). When using two cables for stereo return, the top jack carries the left signal and the bottom jack the right one.

As soon as an audio signal is connected, it is routed to the stereo return: each *Return* section is equipped with its own stereo VCA, which is identical to the ones used in the red, yellow, and green sections of the *Channel* module. To control the amplitude of the effect return, use the Return Level knob (B.7).

The return level can be externally controlled through the CV inputs (B.8), which accept signals with a range of 0÷5V or 0÷10V, allowing any creative use of the stereo effect returns. When a CV is patched to any of these inputs, then the *Level* pot (B.7) acts as an attenuator.

Practice the *Send/Return* modulation with this Technique:
Sidechain #1

5.3 Group Output

Every group has two *Group Outputs* that are stereo paired (A.14, A.15). They can be configured to output just the sum of the processed signals (from the stereo returns) or the overall sum of the channels and effects.

5.3.1 Group Jumpers Configurations

The Group L/R outputs (A.14, A.15) can be configured in two ways through the jumpers on the back of the PCB.

In the first mode, they output just the sum of the two effect returns: this mode is especially useful for further processing of the effects.

In the second mode, they output the same signal that is routed to the Master module: a sum of all the channels, plus the yellow and green effect return. This mode is especially useful for smaller setups, where a Master module is not needed because it allows you to use the Group module as your final mixing unit.

To set the modules to the first mode, place the two jumpers on the back of the PCB to pins 1 and 2 from the left (Figure 12).

To set the modules to the second mode, place the two jumpers on the back of the PCB to pins 2 and 3 from the left (Figure 13).

The signal routed to the Master module via IDC cable will not be affected by any of these jumper configurations.

Figure 12: Pins 1 and 2.

Figure 13: Pins 2 and 3.

It is imperative not to connect the jumpers to any other connector except the two mentioned in this section and shown in the pictures below. Frap Tools may not be held responsible in any way for problems or damage to persons or property or to the device itself if the device is not connected as indicated above.

5.4 Creative Functions

Every Group has three creative controls: the *Group Mute* button (C.1), for all the connected Channels and send/return signals; the *Safe Solo* switch (C.3), which enables the *Solo In Place* function on each of the connected Channels when moved to the left, and the *Stereo PFL* (C.4), accessible via the *Master* module for all the connected *Channels* and *Send/Return* signals.

5.4.1 Group Mute

The Group Mute button (C.1) shuts the whole group, including the effect sends and returns. It equals turning the Group Fader (A.16) all the way to zero.

5.4.2 Safe Solo

The *Safe Solo* function allows the musician to isolate certain channels in the mix, thus muting all the remaining ones. It is achieved through two operations: pushing the *Solo in Place* button on the *Channel* module and moving the Safe Solo switch on the *Group* module: the *Solo in Place* button will determine which channels will be soloed (which will be marked with the white LED) and which ones muted; the *Safe Solo* switch will put this selection into practice (and the group too will display a white LED). It is called *Safe Solo* because it safely allows you to select the channels you want to solo in advance, without affecting the ongoing performance.

5.4.3 PFL

The *Stereo PFL* button (C.4) allows you to prelisten the group when it is muted or when its fader is set to the lowest position. To take advantage of this function, you need to connect the group to the Master module with its PFL Blend (C.5, see below §).

6 MASTER

The Master module is the last step in a CGM setup, and its main purpose is delivering the audio of the Eurorack system to the outside world.

It consists of three sections: the amplitude control (white), the Headphones/PFL section (blue), and the Auxiliary input (purple).

6.1 Amplitude Control

The main control is the *Master Fader* (A.19), which sets the volume of the sound at the Master's left and right outputs (A.17, A.18). This is the overall amplitude of the sum of the connected group, which, at this stage, can gain an additional 6dB.

The Master features two columns of five LEDs each (three green, one yellow, and one red) that act as a visual monitor of the audio levels on the left and right channel, respectively. It acts a LED VU-meter, with the LED values being, from bottom to top: -6dBu, −1dBu, +4dBu, +7dBu, +10dBu.

6.2 Auxiliary Stereo Input

The Master module's purple section features a stereo 3.5 mm jack input (A.20) with a dedicated amplitude control knob (A.22). Patching any audio source to this input will add it to the main mix.

The Auxiliary Input features a PFL button (C.4) just like the Channels and the Groups which allows you to monitor the sound before blending it to the mix.

6.3 Creative Functions

The Master module features a *Headphones* section that is designed for two purposes: 1) monitoring the mix through a pair of headphones, with independent level, and 2) listening to individual channels without them being mixed, through the Pre-Fader Listening controls.

6.3.1 Headphones

The headphones must be connected to the stereo 3.5 mm jack output (A.23). The bottom potentiometer (A.24) sets the volume of the headphones, and the top one (C.5) blends the main out signal (white) with the *PFL* signal (blue).

6.3.2 PFL

PFL (*PreFader Listening*) allows the musician to send to the *Headphones* output any channel, group or the *Aux* input when their main fader is at 0, in order, for instance, to preview it before it is mixed.

PFL is achieved through two operations: pushing the *PFL* button (C.4) on the Channel or Group modules or on the *Aux* input (the blue LED will light up) and adjusting the *PFL* level through the *PFL Blend* knob (C.5).

PFL previews the Channel's sound before the main fader: this means that its amplitude will be determined by the Red VCA level, which means that the green and yellow *Sends* cannot be *PFL*'d from the *Channel*.

This is when the Group *PFL* comes in handy: in order to pre-listen the effect sends and returns before sending them in the mix, you can mute the Group's and activate its *PFL* button: from now on, everything that is managed by the Group will be safely pre-listened without affecting the main mix, including individual Channels' Main Fader settings, sends and returns.

Please note that *Safe Solo* overrides *PFL*: this means that you can perform *Safe Solo* when a Group is in *PFL*, but you cannot *PFL* a channel whose Solo in Place button is off when its Group is in *Safe Solo* mode: the reason is that, as explained above, the *Safe Solo* works as a "*Mute* region," i.e., muting all the unselected channel's red VCAs.

7 TECHNICAL DATA

7.1 Flow Charts

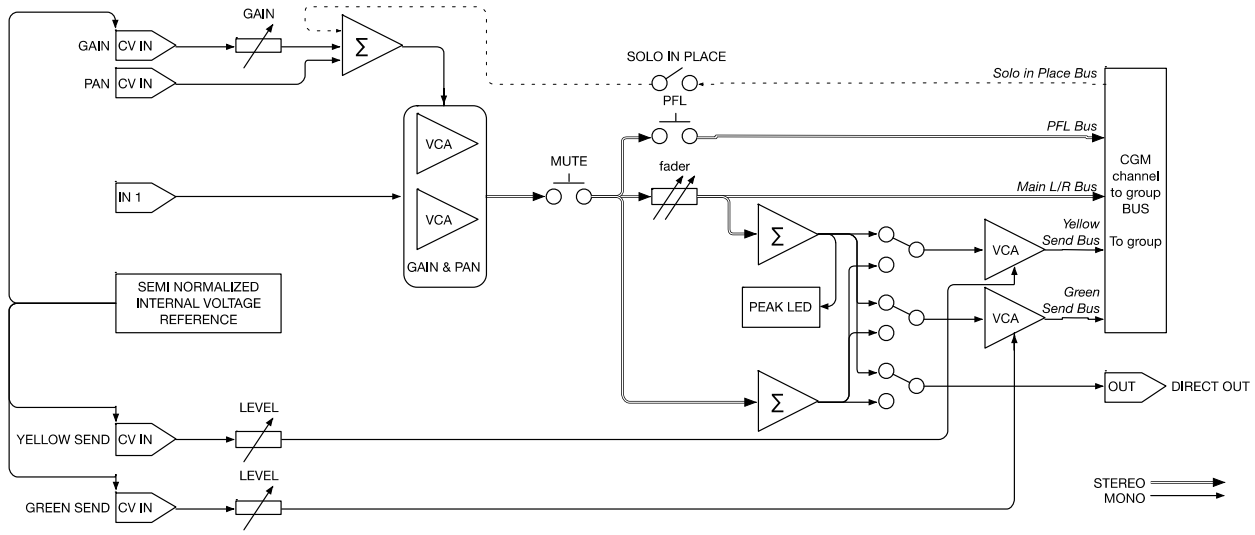


Figure 14: *Channel* flow chart.

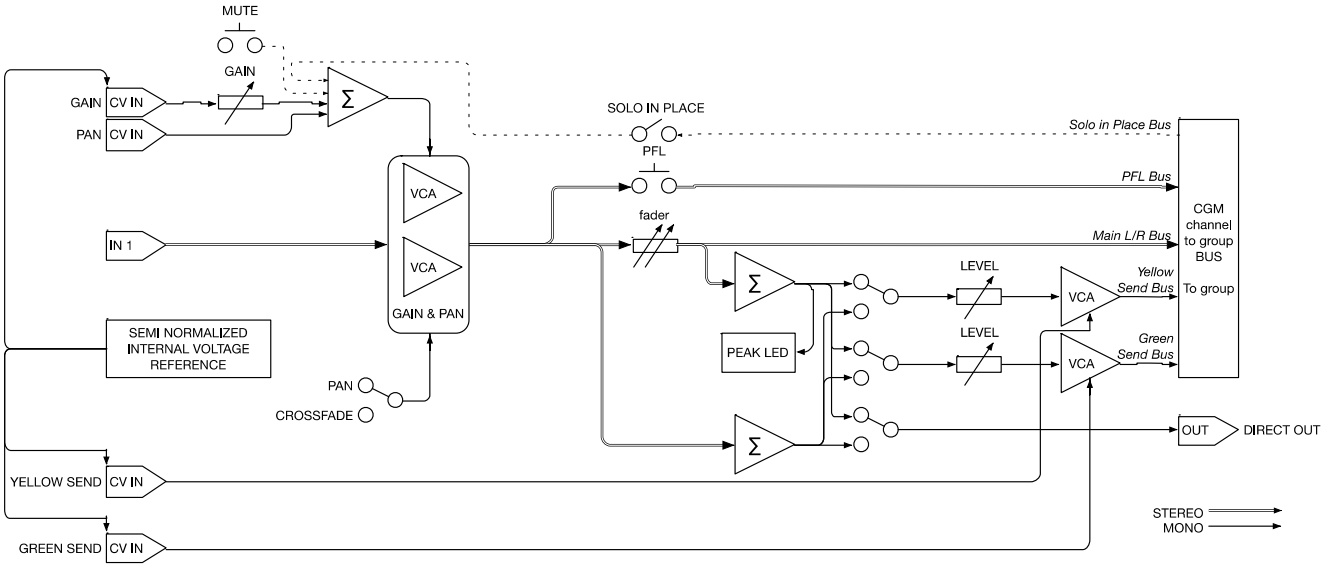


Figure 15: Quad Stereo Channel simplified flow chart (one of the four sections).

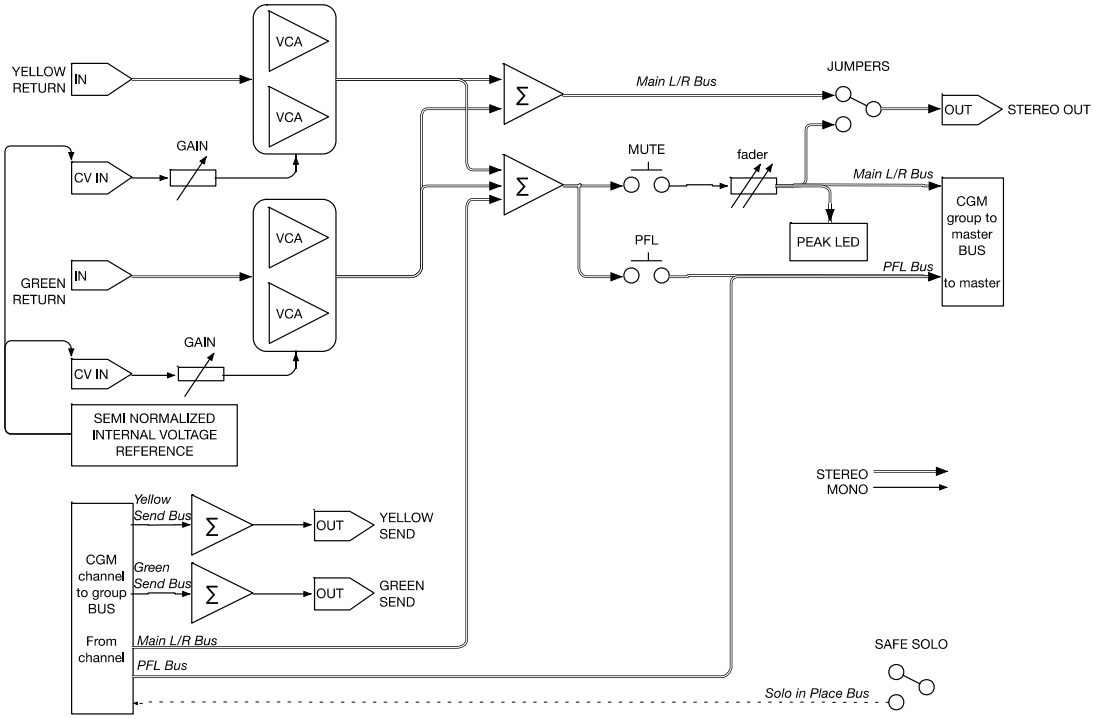


Figure 16: *Group* flow chart.

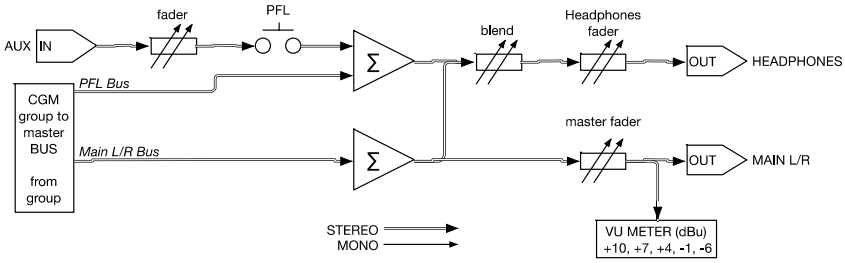


Figure 17: *Master* flow chart.

7.2 SPECIFICATIONS

Parameter	Details		Min	Typ	Max	Unit
Current Draw (0)	Channel	+12V		80	104	mA
		-12V		80	100	
	Stereo Quad Channel	+12		170	200	
		-12		170	200	
	Group	+12V		80	120	
		-12V		75	115	
	Master	+12V		62	101	
		-12V		44	85	
Size	Channel			6		HP
	Stereo Quad Channel			18		
	Group			6		
	Master			6		
Audio Input Impedance	Channel			>25		KΩ
	Stereo Quad Channel	Mono		>10		
		Stereo		>20		
		Aux		10		
	Group	Mono		>20		
		Stereo		>40		
	Master (2)			>3		
CV Input Impedance	Channel			>90		KΩ
	Stereo Quad Channel			>90		
	Group			>90		
	Master			>90		
Gain scale	Channel direct output (3)			7,5		dB
	Send output (3)(4)			4		
	Group output (3)(5)			12,5		
	Master outputs (3)			18		
	Headphones (3)			20		
	Group output (5)(6)			9		
	Master aux input (7)			27		
Frequency response (8)			0,01		40	KHz
Harmonic Distortion (9)				<0.3		%

(1) Max data obtained with heavy driven output onto low impedance load.
(2) Stereo signal on a single stereo jack.
(3) Without any CV applied and all pot at maximum, gain scale, referred to channel input. Measures subject to ±2 dB tolerance.
(4) With 5V CV patched to main VCA.
(5) Jumpers allowing output of sum of channels and return.
(6) Without any CV applied and all pot at maximum, gain scale, referred to group return inputs. Measures subject to ±2 dB tolerance.
(7) Referred to master outputs. Measures subject to ±2 dB tolerance.
(8) Within 1dB, measured at +10dBu at master output onto a 1KΩ load.
(9) THD+N. On a complete system (Channel to group to master) measured at 1KHz with output signal of +10 dBu onto 1KΩ load with a 10Hz to 80KHz bandwidth measurement system.

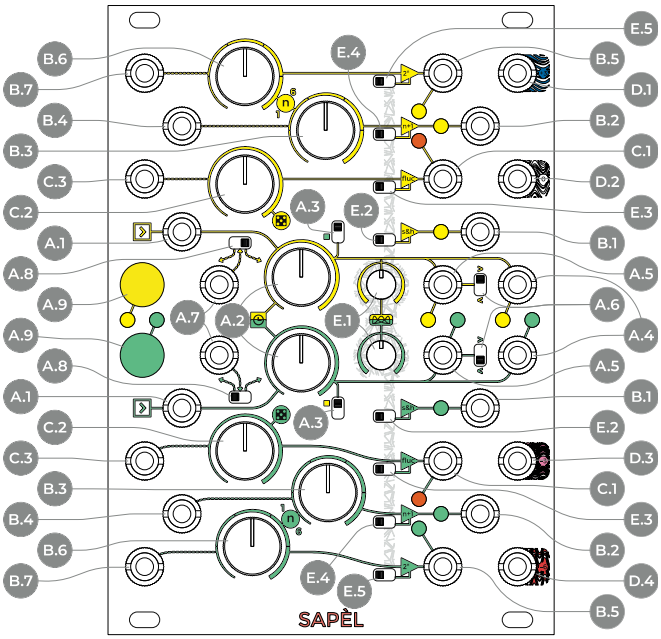


Figure 18: SAPÈL interface.

A **Clocks**
A.1 Clock Input
A.2 Clock Rate
A.3 Single/Both Switch
A.4 Main Clock Output
A.5 Random Clock Output
A.6 Random Clock Mode
A.7 Gate/CV Modulation Input
A.8 Gate/CV Modulation Switch
A.9 Sample-and-Hold (S&H) Button
B **Stepped Random Voltages**
B.1 S&H Output
B.2 n+1 Output
B.3 n+1 Value
B.4 n+1 Value CV Input
B.5 2^n Output
B.6 2^n Value
B.7 2^n Value CV Input
C **Fluctuating Random Voltages (FRV)**
C.1 FRV Output
C.2 FRV Rate and Global Rate of Change
C.3 FRV Rate and Global Rate of Change CV Input
D **Noise Outputs**
D.1 Blue Noise Output
D.2 White Noise Output
D.3 Pink Noise Output
D.4 Red Noise Output
E **Probability Distribution**
E.1 Probability Distribution
E.2 S&H Probability Distribution Switch
E.3 Fluctuating Probability Distribution Switch
E.4 n+1 Probability Distribution Switch
E.5 2^n Probability Distribution Switch

1 PHILOSOPHY AND DESIGN

SAPÈL is an analog generator of random control voltages for Eurorack modular systems. It generates a wide variety of random values to add dynamics to your patches. Such values are sampled on analog thermal noise, to guarantee the truest random generation.

Its main section is composed of two identical clusters of four Sample and Hold circuits each (yellow and green), which generate as many different random values at the same time.

Three of the four S&H circuits in each cluster provide stepped random voltages: two are quantized (in "notes") and one is unquantized; the last S&H circuit features an integrator to generate a fluctuating stream of random voltages.

The three stepped voltage generators are synced, which means that they will output three different values at the same time. The yellow and green generators have two independent internal clocks, which can be replaced with an external one, or temporarily overridden either via external gate or manual button. A copy of the trig used to sample the stepped random values is available, as well as a random trig output.

The fluctuating voltage generator, on the other hand, is independent and has its own potentiometer to define its rate.

Each of the two S&H clusters samples its values from analog noise, thus providing a "true" and completely unpredictable randomness.

The second section of SAPÈL features four analog noise outputs, which are derived from the analog noise used to sample random values.

2 NOISE OUTPUTS

This section features four analog noise outputs, which are, from top to bottom:

- D.1 *Blue Noise* (+3dB/oct spectrum);
- D.2 *White Noise* (0dB/oct spectrum);
- D.3 *Pink Noise* (-3dB/oct spectrum);
- D.4 *Red Noise*, also known as brown or Brownian (-6dB/oct spectrum).

Each noise "color" has its own distinct tone, which can be used for sound-designing purposes.

Theoretically, White Noise has an equal distribution of intensity across all the frequencies per bandwidth. Practically, in the analog domain, White Noise is a sound which has a flat spectrum in the audible range, or, in simpler terms, which can "hit" all the frequencies with equal amplitude at the same time. It is perceived as a highly inharmonic sound but, to the peculiar nature of the human ear, it appears slightly unbalanced towards higher frequencies.

For this purpose, SAPÈL features also a Pink Noise output. Pink Noise is basically a White Noise filtered through a −3dB/Oct filter, which "smoothens" its higher frequencies in order to deliver a more "balanced" sound for the human ear. Pink Noise features an equal distribution of intensity per octave, instead of bandwidth.

Red Noise is similarly generated, but the filter has a slope of −6dB per octave. The result is a low, "rumbling" tone.

Finally, Blue Noise is a kind of inharmonic sound whose intensity increases proportionally to the frequencies. In other words, the higher part of the spectrum will appear to be louder, and the overall result is a high-pitched hissing sound which lacks bass frequencies.

Practice the *Noise Outputs* with these Techniques:
Percussion Sounds #1
Percussion Sounds #2
Percussion Sounds #3
Accents #2

3 VOLTAGE SAMPLING

The yellow and green generators, as said above, work independently the one from the other. Each of them samples three random values at the same time: one which is unquantized (i.e. with no fixed "pitch"), one quantized in semitones and one quantized in octaves.

The sampling process happens when a trig or gate activates the S&H circuit, which "picks" the value played by the noise source at a given time, and "holds" it until another trig or gate is generated. There are four ways to activate the S&H circuit for each of the two generators: internal clock, external clock, manual S&H button, external S&H gate, plus an "extra" mode which combines the clocks (let it be internal or external) of both the green and yellow generators.

By default, each generator is driven by an internal clock. The clock section generates a regular clock signal (called *Main Clock*) and two *Random Clocks* (on which see below). Both the yellow and the green random sources have their own independent clock. The regular train of impulses provided by the *Main Clock* is used to sample the random values and is also routed to the *Main Clock* output (A.4). The *Random Clock* does not affect the S&H circuit, but it is available for advanced modulation purposes through the *Random Clock* output (A.5).

3.1 INTERNAL CLOCK AND CLOCK MODULATION

The built-in clock frequency is managed by the *Clock Rate* knob in the center of the module (A.1). Rotate the knob counterclockwise to decrease the sampling rate and clockwise to increase it.

It is possible to modulate the frequency of the clock via the Gate/CV Modulation Input (A.7).

This input can have two separate functions, selectable via the dedicated switch (A.8): it can route the incoming CV to modulate the clock frequency, or it can use any voltage higher than 3V to trig the S&H cluster (see below, §0).

When such a switch is set to the right, the incoming CV will modulate the clock frequency: a positive CV will increase the clock speed, and a negative one will decrease it.

3.2 External Clock

It is possible to use any external trig to activate the S&H by patching it to the External Clock Input (A.1).

Whenever a cable is patched to this input, the internal clock is bypassed (i.e. it will no longer trig the S&H circuit). This input welcomes trigs and gates only: this means that in order to sample a value, it needs an incoming signal with a really steep rising edge, such as square waves, pulse waves or sawtooth waves with negative ramp (beside of course trig and gate impulses). Other kind of impulses such as sine or triangle waves will be ignored.

3.3 Clock Mix

It has been said that the green and yellow sections of SAPÈL have their own independent clock generation. It is possible, however, to blend them in a more creative way through the *Single/Both* switch (A.3), which feeds the other generator's clock into the one currently in use. To activate the clock mix, set the switch to the position marked by a square of the other generator's color.

This feature works with both internal or external clocks, and it affects the sampling section only: all the clock outputs will maintain their regular behavior.

3.4 Manual Sampling

No matter if you are using the built-in clock or an external one, that stream can be temporarily bypassed with the manual S&H Button (A.9).

By pushing the S&H button the stream of impulses is overridden by a gate high signal, which samples a value and holds it until it is released. A dedicated LED will light up as long as the button is pushed. The main clock output, however, will still output a trig signal.

Practice some uses of the buttons with this *Technique*:
Hold

3.5 External Gate Sampling

This last operation can be automated using the *Gate/CV Input* (A.7).

When the switch (A.8) is set to the leftmost position, any CV signal higher than 3V can be patched in the *Gate/CV* Input and used to override the internal clock.

With this configuration it is possible to use other signal than gates and trigs to drive the S&H cluster, such as sine or triangle waves or even the internal fluctuating random, however a gate or a square signal generally provide best results.

3.6 Clock Outputs (Main and Random)

Every time the S&H Cluster samples a value, a 2ms trig is outputted from the *Main Clock output* (23A.4). If a steady pulse is used, such as the internal clock or an external one, this output will provide an exact copy of the clock.

On the left of the *Main Clock output* lies the *Random Clock output* (A.5), which can either add or subtract trigs from the one in use through its switch (A.6), located between the two clock outputs.

When the switch is up, or in additive mode, it outputs all the clock impulses generated from the clock with the addition of other random clocks; when the switch is down, or in subtractive mode, it randomly subtracts trigs from the ones generated to trig the S&H Cluster, i.e. it outputs only some of the trigs that are outputted by the *Main Clock out*.

In both modes, the random clock density depends on the *Global Rate of Change* (See below, §4.3).

Practice the clocks with these Techniques:
Random Clocks #1
Random Clocks #2
Ratcheting-Like Effect #3
Clock Bursts #1
Clock Bursts #2
Since SAPÈL's clock is 2 milliseconds long, it works really well if patched to FUMANA's *Modulation Input*:
Percussion Sounds #2

4 RANDOM VOLTAGES

SAPÈL is designed to provide a vast array of random voltages with different articulations at the same time. Each of the two S&H clusters (yellow and green) can be divided into two units: the first one generates three random voltages simultaneously, and it is activated by the clocks or gates described above; the second one generates continuous, fluctuating random values and it is completely independent from the clocks and gates.

4.1 Non-Quantized Random Voltages

The most basic stepped random values generator is the Sample and Hold circuit, which outputs its values through the *S&H* output (B.1)

This generator is designed with an independent random generator and creates non-quantized stepped voltages with a range varying from 0 to 7.5V. Non-quantized means that if the values are used, for example, to modify the pitch of an oscillator, the result will be a series of sounds whose frequency may not sit within the conventional 12-semitone Western chromatic scale. It can be used for more experimental music compositions or, more traditionally, to modulate other non-melodic parameters such as timbre, filter frequency, amplitude…

4.2 Quantized Random Voltages

The other two stepped random voltage generators output voltages which are quantized (i.e., "forced") to the 1V/octave standard. If applied to an oscillator's frequency, the result will be a series of random "notes"

The design of these two generators follows the historical Buchla module Source of Uncertainty Model 266, but with a substantial different approach. The circuit has been designed from scratch in order to obtain a more "random" voltages distribution and an extremely precise voltages quantization, capable of generating precise semitones or octaves.

At first glance, the main difference with the *S&H* circuit mentioned above is that the quantized random voltage generator features a knob which controls the n parameter and whose range goes from 1 to 6, as in the original 266 module. The role of the n parameter varies according to each generator's label: 2^n and *n+1*.

The 2^n *Output* (B.5) is quantized in 1/12V steps, or semitones in the 1V/oct scale. In this case, the *Value* knob sets the exponent of 2 which, in turn, determines the number of different values that may be generated by the circuit. Given that *n* can be any number from 1 to 6, there are 6 possible ranges of values that this circuit can generate:

n Knob value	Number of generated voltages
2^1	2
2^2	4
2^3	8
2^4	16
2^5	32
2^6	64

Table 1: *n* values for 2^n quantized random voltages.

Please note that higher is the number, the larger becomes the range of voltages (or "notes") that are generated, starting from 1 (0V) up to 64 (5.25V). This will guarantee the musician more control over the final output and will lead to more expressive results: for example, a low *n* value will always generate smaller intervals and low pitches, while a high one may provide larger leaps from one semitone to the other, as well as higher notes. Please refer to the graph below (Figure 19) for a graphic representation of the exponential note distribution across all the *n* settings:

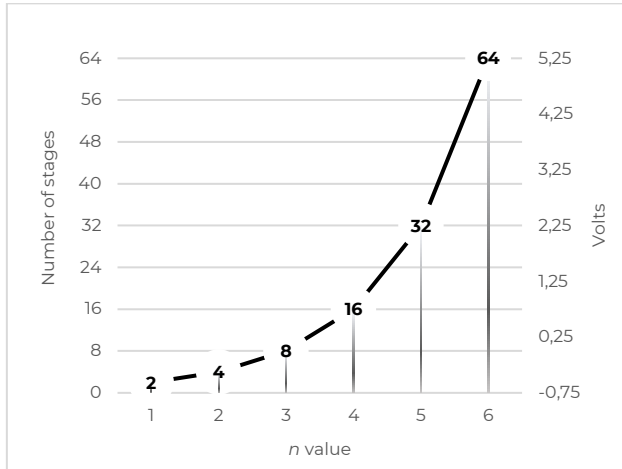


Figure 19: 2^n quantization chart.

The *n+1 output* (B.2) is quantized in 1V steps, or octaves in the 1V/oct scale. In this case, the *Value* knob (B.3) sets the number which will be summed to 1, which, in turn, determines the number of different octaves that may be generated by the circuit. Again, given that *n* can be any number from 1 to 6, there are 6 possible ranges of octaves that this circuit can generate:

n Knob value	Number of voltages generated
1+1	2
2+1	3
3+1	4
4+1	5
5+1	6
6+1	7

Table 2: *n* values for n+1 quantized random voltages.

Even in this case, that higher is the number, the larger becomes the range of voltages (or "octaves") that are generated, starting from 1 (0V) up to 7 (6V). Please refer to the graph below (Figure 20) which displays the linear increment of the octaves across the different *n* settings.

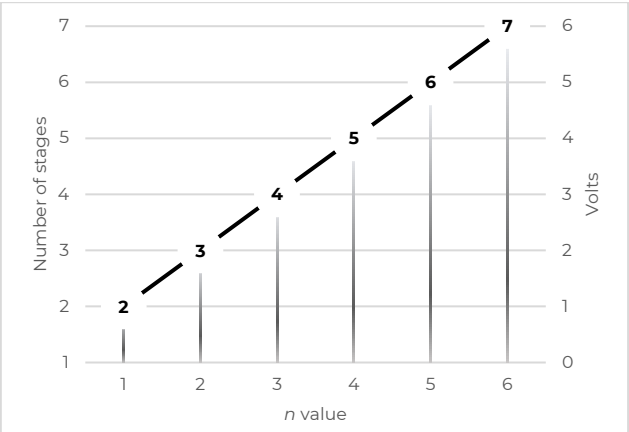


Figure 20: *n*+1 quantization chart.

Both the *2n* and the *n+1* can be controlled via external CV (B.4, B.7) thus allowing the musician to automatically vary the range of values to be outputted.

Practice the use of the *n+1* output with these Techniques:
Ratcheting-Like Effect #2
Voice Spread #1

4.3 Fluctuating Random Output and Global Rate of Change (Random Clock Density Control)

The main purpose of this section is to output (C.1) a continuous, fluctuating random voltage which ranges 0 to 7.5V and whose rate of change (or "frequency") is controlled by its potentiometer (C.2).

This random generator is the only one (in both the S&H clusters) which is not affected by the main clocks or gates; however, on the other hand, it can affect the clock generation itself.

The second purpose of this section, is in fact, to control the *rate of change* of the random clocks: by rotating the C.2 knob clockwise, both the fluctuating voltage frequency and the random clock density (let it be in *more than* or *less than* mode) are increased, and vice versa.

Just like for the quantized voltage generators, this parameter can be modulated with any CV using the its CV input (C.3). The external modulation will affect both the fluctuation rate and the random clock density.

Practice the *Global Rate of Change* with this Technique:
Fluctuating Random & Global Rate of Change

5 PROBABILITY DISTRIBUTION (STORED RANDOM VOLTAGES)

The four S&H generators can be controlled as for magnitude of the voltages that are more likely to be generated: this parameter is called probability distribution, it is globally set by the *Probability Distribution* knob (E.1) and it can be activated independently per each of the four S&H circuits through the four Probability Distribution Switches (E.2, E.3, E.4, E.5).

The knob sets the magnitude of voltages which will be generated more frequently: by default, it is set to the middle position, which means that medium voltages will be outputted more often than high or low ones. Rotate it to the left to raise the probability of generating lower voltages, and to the right for higher ones.

Please note that this setting will not block the SH& circuits to generate different voltages than the ones whose probability is set to be higher: they will just be generated less frequently.

This is a significant difference, for example, from the n Knobs function of the quantized random voltages: in that case, the value defines the "pool" of values that may be picked; here, on the contrary, the Probability Distribution Knob sets the magnitude of voltages that are more likely to be generated within the "pool" selected above.

6 TECHNICAL DATA

6.1 FLOW CHART

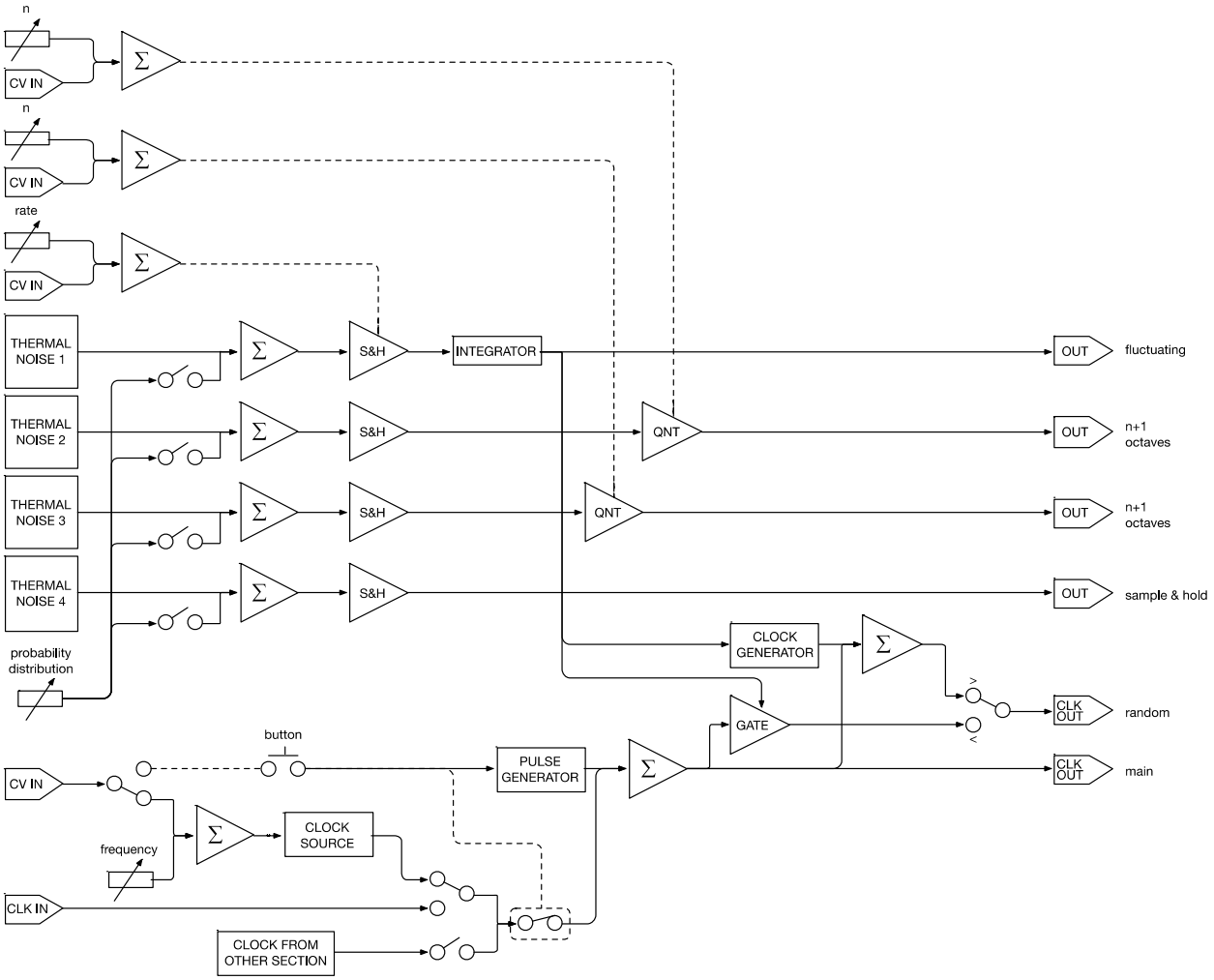


Figure 21: *SAPÈL*'s flow chart.

6.2 SPECIFICATIONS

Parameter	Details		Min	Typ	Max	Unit
Current Draw		+12V			250	mA
		-12V			170	
Size				18		HP
CV input impedance				>90		KΩ
Clock input impedance		on positive pulses		>90		KΩ
		clamping negative pulses		~30		
Built-in clock frequency (1)			~0.1		60	Hz
Quantized Random Voltage Tolerance				<1		%
		global offset	-10		10	mV
Clock output		period		~2		ms
		amplitude		~9		V
Sampling Glitch		on sample & hold			<400	µs
		quantized outputs			n.a.	
Colored Noise Output		blue		10		dBU RMS
		white		10		
		pink		7		
		red		4		
		global tolerance	-2		2	dB

(1) Lower frequencies may be achieved (longer times) with a negative CV used as modulation.

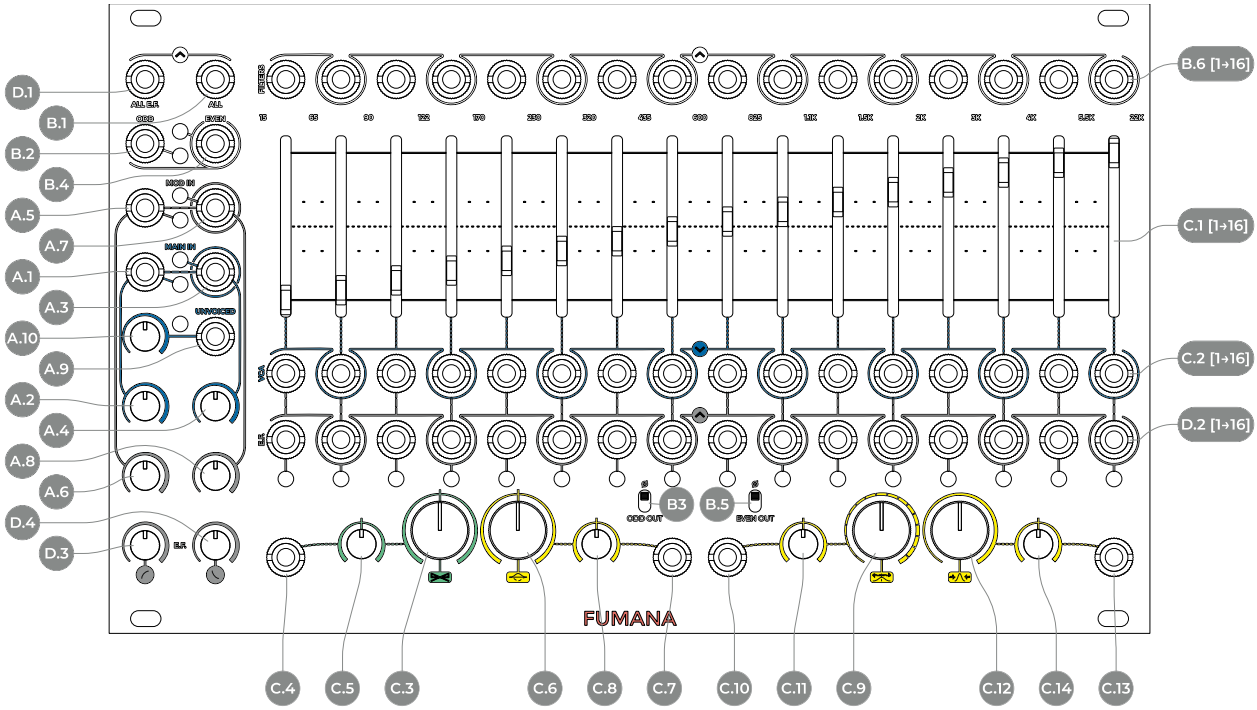


Figure 22: FUMANA interface.

A Audio Inputs
A.1 Main Odd Input
A.2 Main Odd Level
A.3 Main Even Input
A.4 Main Even Level
A.5 Mod Odd Input
A.6 Mod Odd Level
A.7 Mod Even Input
A.8 Mod Even Level
A.9 Unvoiced Input
A.10 Unvoiced Level
B Audio Outputs
B.1 All Output
B.2 Odd Output
B.3 Odd Output Phase
B.4 Even Output
B.5 Even Output Phase
B.6 Individual Band Outputs
C Controls and CV Inputs
C.1 VCA Faders
C.2 VCA CV Inputs
C.3 Tilt Control
C.4 Tilt CV Input
C.5 Tilt CV Input Attenuverter
C.6 Peak/Notch Control
C.7 Peak/Notch CV Input
C.8 Peak/Notch CV Input Attenuverter
C.9 Center Control
C.10 Center CV Input
C.11 Center CV Input Attenuverter
C.12 Width Control
C.13 Width CV Input
C.14 Width CV Input Attenuverter
D Controls and CV Outputs
D.1 All Envelope Followers Output
D.2 Individual Envelope Followers Outputs
D.3 Envelope Followers Rise Control
D.4 Envelope Followers Fall Control

1 PHILOSOPHY AND DESIGN

FUMANA is a dual all-analog fixed filter bank. Each filter bank is composed of an array of 16 independent bandpass filters tuned to specific frequencies.

FUMANA is designed around one basic core principle: modify the spectral content of the incoming audio signal by filtering it through 16 bandpass filters in parallel and then vary the amplitude of each resulting band.

Even though this core principle is relatively simple, the key feature of FUMANA is that it provides a wide set of controls over the bands' amplitude: the spectral content of the incoming sound can be thus modified by the faders, which are individually CV-controllable; by the envelope followers which are generated by the analysis of the sound patched in the Modulation input; by the global parametric controls such as *Tilt* and *Scan*.

Furthermore, a flexible input/output signal routing allows the musician to "split" the 16 filters into two 8-band spectral processors by grouping the odd and even bands separately. It will be possible to process two independent signals, or to blend two signals into one output, or even to perform two different sonic treatments over the same signal and route it to two different output sections.

Individual outputs are available for each band, both for the Main sound (i.e. the filtered one) and for the Mod sound (i.e. the resulting envelopes).

Since a spectral transfer tool may be used as a "vocoder-like-effect", the FUMANA provides an input for an external noise which may be used for fricative/sibilant sounds.

1.1 Spectral Transfer: A Brief History

In the modular synth domain, the most famous device for performing spectral transfer was the legendary Model 296, designed by Donald Buchla in the seventies. It was a 16 bands equalizer, which featured one input for the odd bands and one for the even bands: this solution allowed the artist to process two different signals at the same time, but it also allowed to transfer the harmonic content of one to the other via dedicated switches. It was thus possible to analyze the odd bands and transfer their amplitude to the even bands, and vice versa.

This design was clever because it took advantage of a single array of 16 filters, but the resulting spectral transfer was somehow "approximated" because the modulation signal came from the analysis of the adjacent band, i.e. another frequency area of the spectrum, albeit quite close.

The FUMANA pushes this concept even further, but still within the analog domain. An additional bank of 16 filters is added, specifically for analysis purposes, with a different band-pass slope (see below §5).

When performing a spectral transfer, it is fundamental to consider the harmonic content in both main and modulation audio signals. Remember that if there is no harmonic to excite the content on the main signal, there will not be that much outcoming audio, even if all the VCAs are opened at their maximum level:

Poor harmonic content on the *Main* signal: if a pure sine (one harmonic) or any signals with poor harmonic content is used, there are not many chances to hear good results exciting several bands. Supposing it is a sine wave with a frequency of 105Hz (which lies then on band 3), theoretically something happens only when band 3 is excited from the modulation signal.

Rich harmonic content on *Main* signal: this means more chances to have several bands excited by the envelope followers control voltages created by the modulator filterbank.

Poor harmonic content on *Modulator* signal: allows creation of very selective envelope followers signals, which translates in an extremely selective spectral transfer. If the modulation signal changes pitch or has a very variable content, it will for sure result in a more heterogeneous spectral transfer.

Rich harmonic content on *Modulator* signal: allows a very rich spectral transfer. Just keep in mind that if you are using a square wave in the modulation with frequencies varying from 100 to 200 Hz, probably you won't hear that much difference, due to the richness of harmonics of the square wave.

1.2 Panel Overview

A consistent color and graphic coding make the front panel easy to understand at a glance, once properly understood.

The main graphic solution is the distinction between the odd and even bands: since the two groups can work independently, every input or output related to the even bands is marked by a circle around the jack socket. (The bands are numbered 1-16 from left to right: odd bands are band 1, 3, 5, 7, 9, 11, 13, 15; even bands are 2, 4, 6, 8, 10, 12, 14, 16).

Another key coding is the distinction between the *Main* filter array and the *Mod* filter array. Everything that relates to the *Main* filter bank, which is the circuit directly affecting the sound that is heard, is marked by the color blue, let it be audio output, CV input or even the faders' LED color.

On the other hand, everything that relates to the *Mod* filter bank, which is the one that extracts the envelopes from the modulating signal and modifies the harmonic content of the main filter bank, is marked in grey, including the individual envelope output LEDs (which flash white when active).

Finally, the green and yellow colors mark the Global Spectral Editing tools: the former relates to the Tilt control, the latter to the Scan controls.

2 AUDIO INPUTS

FUMANA has two pairs of inputs, which are called *Main In* (blue), and *Mod In* (gray), short for "modulation input", plus a fifth input called *Unvoiced*. (A.1, A.3, A.5, A.7, A.9).

Each pair is composed of inputs for the *Odd* and *Even* bands, which are semi-normalled together. This means that when only one patch cable is connected to one of the two inputs, it automatically feeds the other. If you want to use different sources for odd and even bands, simply use two different cables. If instead, you want to feed only the odd bands and not the even (or vice versa), simply feed the odd input, and plug a dummy cable to the other input (or vice versa).

Each of the four inputs has its own amplitude potentiometer (A.2, A.4, A.6, A.8): different levels can thus be set for odd or even filters within each pair of input. This also means that if you want to emphasize even bands on the main signal, you can simply add more gain to them, and less to the odd ones.

The fifth input, named *Unvoiced*, is designed for adding some depth to the fricative consonants that might be missed when performing vocoding-like operations. It features its gain control as well (A.10).

The red LEDs connected to the odd and even inputs light up displaying the amplitude of the incoming audio after the gain level.

3 AUDIO OUTPUTS

FUMANA's main audio outputs are the three in the top left area: *All* (B.1) outputs all the bands, while *Odd* (B.2) and *Even* (B.4) output respectively only the odd and only the even bands.

In addition to these, 16 direct outputs are also provided (B.6), which are located on the top of each band. The main difference between these and the other three outputs is that, while the *All*, *Odd* and *Even*, are sums of groups of bands after their respective VCAs, these 16 *band outputs* are pre-VCAs outputs, directly from the bandpass filters.

These are very useful in case it is needed to process in parallel only a single band or a group of selected bands. In that case the 333 module can be very helpful, since it is capable of summing perfectly up to 7 signals into a single jack.

The use of the individual filter outputs does not affect their respective band's presence in the *Odd/Even/All* outputs: these stages are completely independent.

The *Odd* and *Even* outputs also feature a phase inversion switch (B.5): this switch may be useful in case you want to merge one of that signal with the *All* output, and dynamically emphasize (phase summing) or dynamically attenuate (inverted phase summing) for example only the *Even* bands (or *Odd*, depending on needs).

Practice the *Odd/Even* swtich with this Technique:
https://frap.tools/fumana-feedback-1/

The result of the sum of *Odd* and *Even* outputs is slightly different from the *All* output. This is because the *All* output uses an additional low pass filter at 18KHz in order to obtain a less "edgy" upper end, often resulting when using dense signals and/or heavy modulations (you can have a clear view of this looking at the Transfer Function Details). In case you need a crispier sound, consider using the *Odd+Even* combination.

4 AUDIO PROCESSING AND MODULATION PATH

FUMANA's filterbank processes the sound patched to the *Main* input by varying each band's amplitude through a VCA circuit. Such variation can be achieved in four different ways, many of which can easily co-exist:

- through the individual band faders;
- through the individual band CV inputs, right below the faders;
- through the *Macro Spectral Editing* tools (*Tilt* and *Scan*)
- through the spectral transferring function performed by any sound patched to the *Mod input*.

The result of all these modulations is outputted by the *All*, *Odd* and *Even* outputs, and it is visually displayed by the 16 blue LEDs placed on the 16 band faders, whose intensity graphically displays the amplitude of the respective band after any modulation applied. The relationship of the modulations is displayed in Figure 23, and it will be further explained in the next paragraphs.

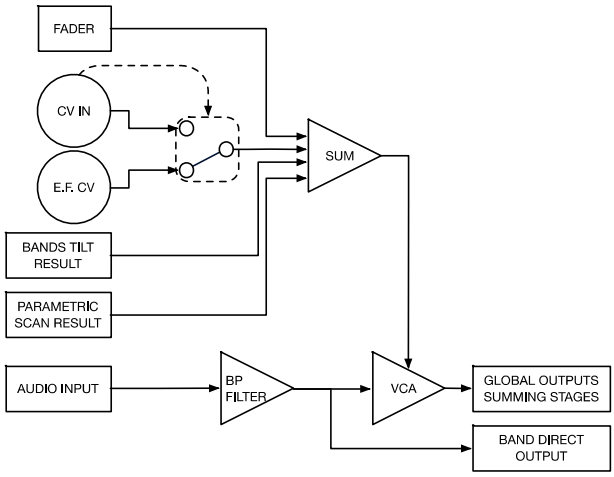


Figure 23: FUMANA's modulation routing.

4.1 Faders and CV

The main CV comes from the 16 faders (C.1), one per band, on the main panel. In the lowest position the VCA is closed: raise the fader to increase the selected band's amplitude. This operation can be automated through the 16 individual jack sockets below each fader (C.2), which provide inputs for external CVs. The CV inputs welcome any signal either bipolar or unipolar, even at audio rate, up to ~1000Hz (after which a lowpass filter is applied, see Figure 24): it is thus possible to perform AM over an individual band!

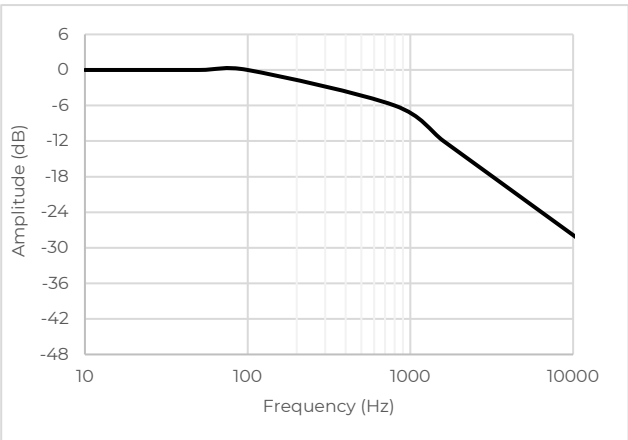


Figure 24: CV frequency roll-off.

4.2 Macro Spectral Editing

The yellow and green areas at the bottom of the module are, respectively, bands parametric scanning and bands tilting. These two functions are totally independent the one of the other. These have been designed to quickly modulate multiple bands with a few controls. Each of these parameters has a main potentiometer (C.3, C.6, C.9, C.12) and its own CV input (C.4, C.7, C.10, C.13) with a dedicated attenuverter (C.5, C.8, C.11, C.14).

4.2.1 Tilt

The green section is the *Tilt* parameter (C.3, C.4, C.5). Assuming a fixed fulcrum in the middle of the 16 bands (between band 8 and 9), the Bands Tilting allows to gradually emphasize half of the bands, as much as the band itself is far from the fulcrum and attenuate the other half in the same way. It is then possible to change the balance between lower or higher frequencies. The parameter has its rest at the center. Moving it counterclockwise the bands from 1 to 8 are emphasized, with an emphasis that gradually decreases from 1 to 8. At the same time, bands from 9 to 16 are progressively attenuated, from band 9 to 16.

One of the possible uses of the band tilt is to use it to temporarily attenuate the lower frequencies' presence when the mix is particularly "crowded" in that frequency range. In fact, feeding the CV with the envelope you use to create your basses, or passing the audio of the bass drum through an envelope follower, results in a temporary emphasis on the higher frequencies. Balance the tilt with its main pot, and simply play with the attenuator and envelope follower times to obtain the desired effect.

4.2.2 Parametric Scanning

The yellow section is called *Parametric Scanning* due to its similarity to the use of 3 variables as the parametric EQ. In any parametric EQ, it is possible to set the center frequency, the gain value, positive or negative, also called peak/notch in some cases, and the slope. Here it is similar, but it doesn't work directly as a filter, but as a voltage control which feeds each band VCA.

The first control is called *Peak/Notch* (C.6, C.7, C.8) and it selects the amount of emphasis or attenuation of the scan: when the knob is at its center, no emphasis is applied; when it is moved clockwise, a positive gain offset is applied; when it is moved counter-clockwise, a negative gain is achieved, which can be useful to perform notch-filter-like operations.

The *Center* control (C.9, C.10, C.11) sets where this emphasis/attenuation has its maximum/minimum level: when fully counterclockwise, no band is emphasized; by rotating it clockwise, the position shifts from band 1 to band 16; when fully clockwise, no band is emphasized as well.

With the *Width* parameter (C.12, C.13, C.14) it is possible to emphasize nearby bands: you can set a width from none to all 16 bands (16 bands are audible only if the *Center* knob is set to noon).

To bypass the parametric scanning simply move the *Width* control completely counterclockwise.

Practice the different parametric scanning controls with these Techniques:
Percussion Sounds #1
Percussion Sounds #3
Raw CV

4.3 Spectral Transferring: Modulation Filters and Envelope Followers

The modulation circuit is designed to perform spectral transfer between the modulation signal and the main one.

The modulating signal must be patched to the *Mod* input, which feeds an array of 16 band-pass filters similar in design to the ones in the main array; then, each filter feeds a dedicated envelope follower; the resulting 16 envelopes create as many different control voltages that are semi-normalled to the VCA input jacks (C.2).

As said above in the paragraph about the Modulation Routing, patching a jack into any individual band CV in will break the semi-normalization of the envelope followers.

The duration of the resulting envelopes can be modified with the *E.F. Attack* and *Release* control knobs (D.3, D.4). The leftmost position offers the fastest envelope response; rotating it clockwise will result in slower envelopes.

Usually, the only parameter in the time domain used to modify the envelope followers' signals is the Release time. FUMANA is equipped with the Attack time as well, in order to have more control over the harmonic content modification and obtain more subtle results.

The circuit response is non-linear, meaning that the knobs allow for more precise control over the fast times rather than the long ones, which is useful when a signal with fast transients is being processed. Furthermore, given that the more conventional use of this circuit is to perform vocoding-like operations, it is more frequent that fastest envelopes are required.

Each envelope uses its own independent time-scaling factor, which is longer for lower frequencies and faster for the higher frequencies, in order not to cut any audio wave semi-period.

The envelopes resulting from this spectral analysis are also available on the 16 *E.F. outputs* (D.2), which can be used as CV to be routed to different points of the patch without affecting their transfer to the VCAs of the main filters.

An *All E.F.* output is also provided, which sums all the envelopes together (D.1).

The amplitude of the Envelope followers also depends on the level of the modulation source: the FUMANA is designed to work with input potentiometers in center position (12 o'clock) when modular levels are used (bipolar 10Vpp). Potentiometers can be used to amplify or attenuate particularly hot signals.

Practice some peculiar spectral transferring patches with these *Techniques*:
Kick Drum #2
Percussion Sounds #2
FUMANA Feedback #3

4.4 The 'Unvoiced' Section

Unvoiced sounds (fricative/sibilants) are common in human languages and can be found in words containing or starting with s, f, z, ch and other fricative sounds ([s] [z] [ʃ] [tʃ] [dʒ] [ts] [ʂ] [f] [v] [ɸ] [θ] [ʒ] etc.).

Historically, all vocoders have an additional section to manage these kinds of sounds. Usually that section worked with a sort of "unvoiced detection circuit": you can get an approximated idea of it imagining a "de-esser", which detects the presence of certain frequencies in the sound spectrum, and instead of doing a selective band compression attenuating that frequencies, it controls a sort of mixer that changes the input of the filter from the main to a noise signal, and is capable of very fast transients.

The FUMANA has been designed to be a spectral editing tool, more than a vocoder: for this reason, since its sketches shared a lot of things with vocoding circuits, we developed our own approach of the *Unvoiced* section, which for sure leads to different results. It works managing the amplitude of the noise and summing the result with the main signal on 2 selected bands, instead of varying the mix between main and noise signal. The two selected bands are the 14 and 15. The jack socket in the *Unvoiced* area is the input (A.9) for the noise audio signal, that may be provided by the SAPÈL (we tend to suggest pink noise or white noise). The potentiometer (A.10) sets the noise level.

It is also important to specify that even if the input is one (mono), the detector and the amplitude manager of that noise is dual and totally independent. Band 14 has its own envelope follower signal controlling its own VCA, as well as band 15. This means that when used with two different modulation signals and two outputs (odd and even), the result of the unvoiced signal added to the main one is totally independent.

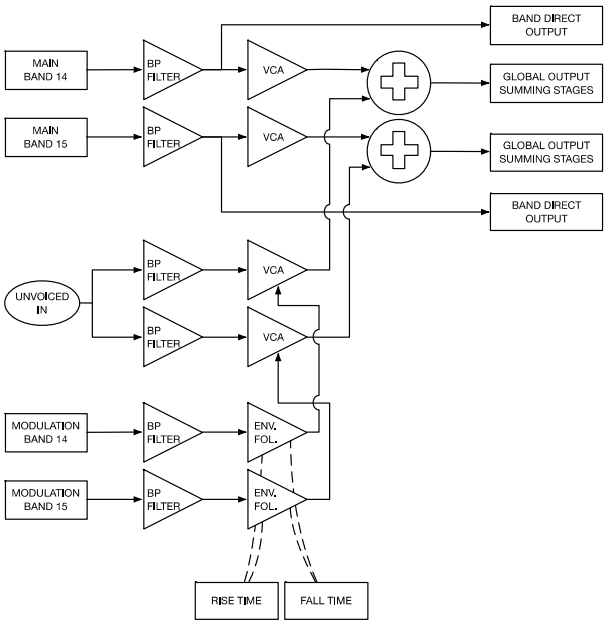


Figure 25: FUMANA's *Unvoiced* section.

Also, we worked on the way these envelopes are achieved: in the FUMANA they tend to be much "softer" than normal to reduce risks of "envelopes bounces" resulting in really fast "noise sparks". This approach allows to use the unvoiced section also for non-vocal purposes. A good example is using drums with cymbals/hats as modulation source: in this case, the unvoiced sections helps to detect when there is material and recreate cymbals or snare wires. Anyway, this does not limit you to put

any kind of waveform in the unvoiced input and start to experiment by yourself.

5 FILTER DESIGN

Each of the two filter arrays is based on 16 parallel analog bandpass filters. The main filter bands from 2 to 15 are mainly based on Bessel calculation, while bands 1 and 16 are respectively a lowpass and highpass filter with a custom method to obtain better musical results. All bands on the main filter array use an 8th order slope (48dB/oct).

Conversely, the modulation filter array uses a 6th order filter slope (36dB/oct), and an additional stage to compensate each band's energy. The calculation method used was optimized for this specific purpose.

The crossover frequency between each filter from band 2 to 15 is an approximation of an interval of ~5.5 semitones (or 11/24 of an octave). This results in a distance between odd bands (and also between each even band) of ~11 semitones (11/12 of an octave), or a Major 7th. The ratio is defined in order to limit the chance to obtain a recursive tone emphasis/attenuation over the whole audio spectrum.

The first and last band's frequencies are calculated in order to achieve the best regulation of "sub-bass" and "upper-end".

The filters are designed to guarantee the flattest frequency response on the ALL output, which within ±1.5dB from band 2 to 15, and globally between ±4.5dB. This achievement translates into an extremely transparent filter.

Each band has an individual trimmer to set its band amplitude in case a different result is needed. The trimming procedure is NOT present in this manual since it MUST be performed by Frap Tools authorized personnel only. Frap Tools may not be held responsible in any way for problems or damage to persons or property or to the device itself if any trimming or disassembly is tried or performed without authorization.

6 PATCH EXAMPLES

The following examples assume the same starting point. The first time you use the FUMANA and you're not 100% familiar with it, we suggest to keep these settings:

· set the 4 volumes of the *Odd* and *Even* bands of both the*Main* and *Modulation* section to the middle (at 12 o'clock);

· keep the envelope followers' *rise* and *fall* times at minimum (counterclockwise);

· put all volume faders to the minimum.

6.1 16-Band Spectral Transfer

The easiest thing to perform a 16 bands spectral transfer is patching a square wave to any of the main input (blue) and a triangle or sine wave to the modulation input (grey). Patch only one cable to the inputs and leave the other unpatched: this automatically feeds the other input thanks to internal semi-normalization. Now patch only one output (ALL output) to the input of your mixer to hear properly what happens.

At this point, if the modulation signal is within the audio range and has a "modular" amplitude level (bipolar 10Vpp), you will notice that at least one of the white LEDs lights up. This means that an envelope follower CV has been created in the modulation filter array and has been transferred to the main filter section VCA of that band.

6.2 Dual 8-Band Spectral Transfer

Now you need 4 signals to do a proper dual 8 bands spectral transfer. For this example, you can proceed with a square and a pink noise into the main inputs (blue). The modulation inputs (grey) may use, only for example purposes, a simple triangle wave on the odd bands and a percussive signal on the even bands. Now patch two outputs (Odd and Even) to two different input of your mixer to hear properly what happens.

At this point, if the modulation signals are within the audio range and have a "modular" amplitude level (bipolar 10Vpp), you'll notice that, when varying the frequency of the triangle wave onto the odd bands, the odd LED will start to light up, and you will hear the result of this spectral transfer via the odd output, while the even bands LEDs will light up based on the amplitude and the harmonic content of the percussive signal.

6.3 Hybrid Spectral Transfer

You can do even a sort of hybrid spectral transfer using, for example, two different signals for the main input and one as a modulator. In that case, if they come both from the same oscillator, and you pick up the output from the All output, you'll obtain a more "complex" signal, which merges together two different waves, in-phase. Otherwise, using different sound sources, and picking up the signals from the Odd and Even outputs, you'll obtain two different signals, which can be processed independently, but which shares a similar harmonic emphasis.

Of course, you can do the opposite, using two modulators on the same single source applied onto the main filter array.

6.4 Vocoder-Like Behavior

The easiest way to perform a 16 bands kind of vocoder is very similar to the 16 bands spectral transfer: you can

patch a square wave, or any signal with rich harmonic content (which may also vary in terms of pitch and timbre), or a noise (i.e. a non-pitched sound) to any of the main input (blue). Then, apply a voice to the modulation input (grey). This automatically translates in a vocoding effect. The use of the unvoiced section may also be of interest.

Having 2 main signals and 2 separates voices translates the FUMANA in a raw dual 8 band vocoder: of course, the results cannot be at the level of the 16 bands, since it is using only half of them.

7 TECHNICAL DATA

7.1 TRANSFER FUNCTION

All tests performed after 30 minutes of warmup at room temperature of 23°C using pink noise at a sampling rate of 96KHz. The resolution used is 65'536 points which corresponds to ~0.732421875Hz of windowing.

Please note that the transfer function of the module may vary from module to module, ±2dB is still accepted. The causes are the components tolerance and the trimming which is done manually per each band to define their amplitude. The trimming is done to achieve the flattest band on the all output. Results on odd or even are then a consequence of this setting.

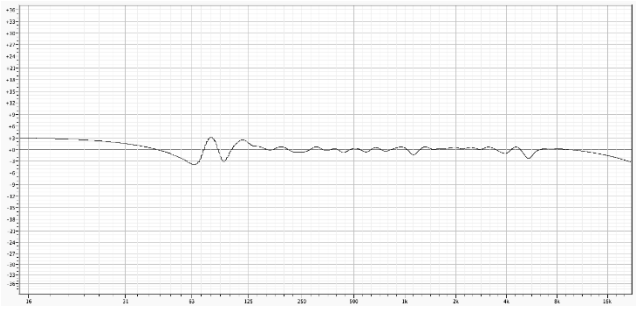


Figure 26: All bands output, complete.

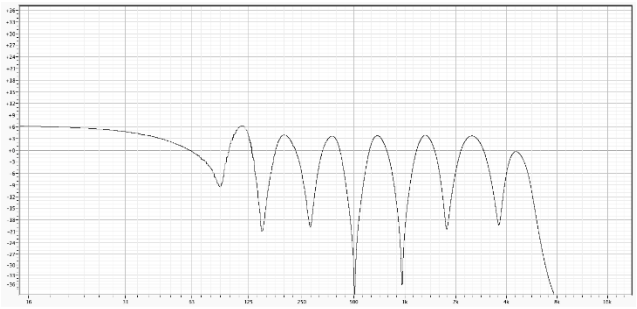


Figure 27: Odd bands output, complete.

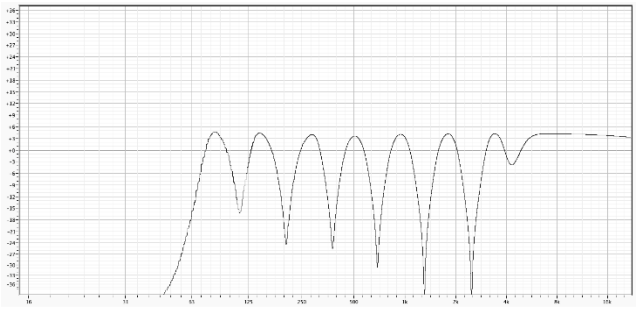


Figure 28: Even bands output, complete.

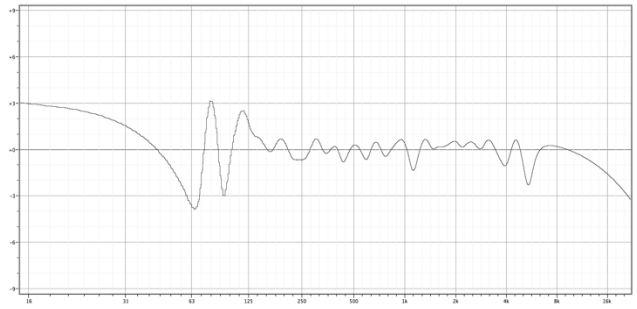


Figure 29: All bands output, detail.

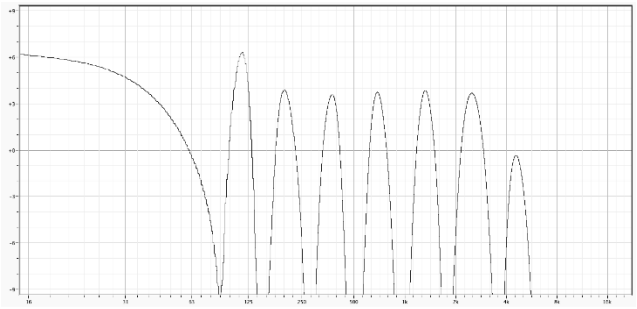


Figure 30: Odd bands output, detail.

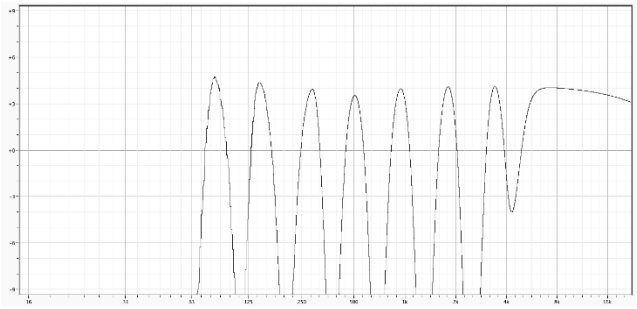


Figure 31: Even bands output, detail.

7.2 SPECIFICATIONS

Parameter	Details		Min	Typ	Max	Unit
Current Draw		+12V			430	mA
		-12V			390	
Size				42		HP
CV input impedance				>90		KΩ
CV output impedance				<50		Ω
Audio Input impedance				>20		KΩ
Audio Output impedance				<50		Ω
		Individual Filters		<250		
Envelope followers' amplitude from 0/10V. (1)				±5		%

(1) To avoid incorrect behavior when driving external VCAs, a slight negative BIAS voltage is applied to each envelope circuitry to prevent a minimum positive voltage on output due to components' tolerances.

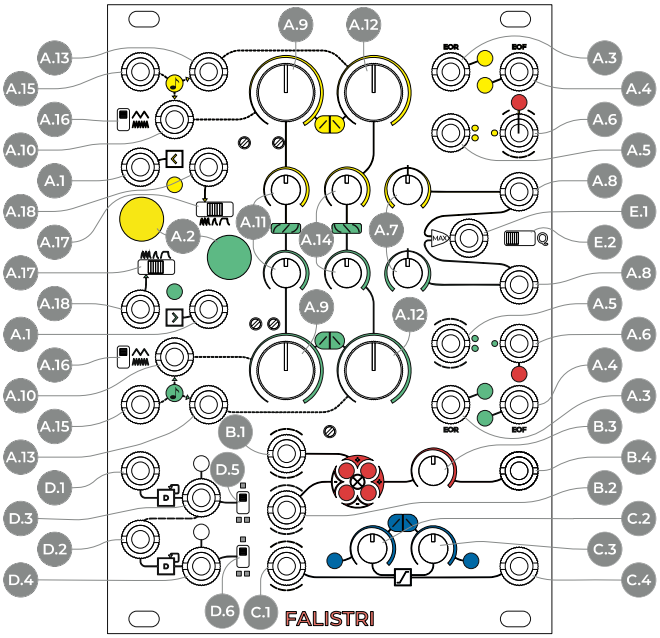


Figure 32: FALISTRI interface.

A **Generators**
A.1 Trig/Gate input
A.2 Trig/Gate Button
A.3 End Of Rise
A.4 End Of Fall
A.5 Bipolar Output
A.6 Unipolar Output
A.7 Attenuverter
A.8 Attenuverted Output
A.9 Rise Time
A.10 Rise CV Input
A.11 Rise Shape
A.12 Fall Time
A.13 Fall CV Input
A.14 Fall Shape
A.15 V/oct Input
A.16 Time Scale
A.17 Play Mode
A.18 Force Loop Input
B **Four-Quadrant Multiplier**
B.1 Four-Quadrant Multiplier Input 1
B.2 Four-Quadrant Multiplier Input 2
B.3 Four-Quadrant Multiplier Level
B.4 Four-Quadrant Multiplier Output
C **Linear Slew Limiter**
C.1 Linear Slew Limiter Input
C.2 Linear Slew Limiter Rise Control
C.3 Linear Slew Limiter Fall Control
C.4 Linear Slew Limiter Output
D **Dual Cascaded Frequency Divider (DCFD)**
D.1 DCFD Input 1
D.2 DCFD Input 2
D.3 DCFD Output 1
D.4 DCFD Output 2
D.5 DCFD Range 1
D.6 DCFD Range 2
E **Quadrature Section**
E.1 Max Output
E.2 Quadrature

1 PHILOSOPHY AND DESIGN

FALISTRI is a fully analog multipurpose movement manager designed to generate and edit voltages to easily accomplish any patch. The module can be divided in two big parts: the upper two-thirds of the panel are occupied by two specular function generators, while the lower third is composed by a dual cascaded frequency divider, a linear slew limiter and a four-quadrant multiplier.

2 FUNCTION GENERATORS

The generators are designed to quickly and intuitively achieve the function needed, providing several outputs and inputs to dynamically change their times and trigs. The main peculiarity in this design is the way shapes are managed:

· times are independent from shapes – when sculpting an envelope, you can freely morph from logarithmic to linear to exponential slope, without changing the time needed to complete *Rise* or *Fall* (and their *Loop* time when used as oscillator);

· shapes are exclusive per each stage – you can have logarithmic *Rise* and logarithmic Fall, Linear Rise and Logarithmic Fall or any other combination of the modes. Of course, any blending in between these shapes is achievable, allowing deep sculptures of sound when used as Control Voltage as well as audio source.

The time switch scales the time factor on both rise and fall stages in each generator.

Both the manual *Buttons* (for hands-on control) and the external *Trig/Gate Inputs* are available, and are used in the three possible play modes:

· *Loop* – where end of fall stage triggers the start of rise, incoming triggers are not needed, but, if present, recall the rising stage;

· *Transient* – only the low-to-high transition of the trigger/gate signal is used to start the rise stage, while fall stage is automatically recalled at the end of rise;

· *Hold* – the low-to-high transition of the trigger/gate signal is used to start the rise stage, while the fall stage is trigger by the high-to-low transition of the *Trig/Gate* signal.

A Gate input to force loop is also included, and it is possible to use *Quadrature* modes to add more complex relationship between yellow and green functions.

Practice some uses of the buttons with this *Technique*:
Hold

2.1 Times

The stage times are managed by the *Rise* and *Fall Time* knobs (A.9, A.12). The *Rise* knob lets you set how much time the function takes to reach the maximum level. The *Fall* knob let you set how much time the function takes to decrease and "rest" after the rising stage (when in *Transient* mode) and/or after hold (when in *Hold*).

These parameters can be independently modulated via any control voltage using the jack sockets connected to them on the left (A.10, A.13). Any positive voltage increases the knob value, while a negative one decreases it.

It is also possible to modulate these parameter simultaneously through the *V/oct input* (A.15), which is connected to the rise cv in and fall cv in. This input works in the opposite way; positive signals decrease the time and negative ones increase it. This socket is marked with a note symbol to point out that it is the one you may want to use as *V/oct Input*.

Keep in mind that to obtain a perfect behavior over various octaves, you may want to keep both the *Rise* and *Fall* stages at similar values to make them use the same portion of: the V/oct converter.

It is possible to choose between two different *time scales* using the switch on the left (A.16): the upper position has longer times, and it is thus more suitable for envelopes and LFOs; the lower position, on the other hand, provides extremely short functions on the microsecond time scale, and when looping it can be used as an audio-rate oscillator. This is just a general reference, as any possible combination of fast/slow and transient/cyclical is at hand for your sonic palette.

Practice the *Rise* and *Fall* CV inputs with these Techniques:
Accents #2
Ratcheting-Like Effect #2
Percussion Sounds #1

2.2 Shapes

Each stage's *shape* can be independently selected. A small potentiometer, below each time pot, allows you to morph between three shapes, from logarithmic to linear to exponential (A.11, A.14). It is extremely important to specify that the change of shape is completely independent from the time factor: this means that you can freely play with different wave shapes without changing the duration of the envelope / LFO, or without affecting the pitch when using FALISTRI as an oscillator. In this last case, the two small pot will play the role of an actual waveshaper.

Practice waveshaping with this *Technique*:
Waveshaping

2.3 Trig and Modes

The switch on the right (A.17) provides you the "play modes", which are, from left to right:

Loop – the function automatically retrigs when it reaches the end of its cycle (use this for an LFO or a classic audio oscillator);

Transient – creates one-shot rise-fall function, or AD envelope;

Hold – when the signal used to trig the function stays "high" (more than 1.5v) for a longer time than the one of the rise stages, the function generated has a "hold" stage, thus being an AHR envelope. The time of "hold" is the difference between the gate time and the rise time. After that, when the gate goes low, the fall stage is recalled.

When the switch is in *Transient* or *Hold* mode, patching a positive voltage into the jack socket will force the play mode to *Loop*. This input welcomes also audio-rate signals, for more experimental effects.

If a gate-high signal is patched to the jack connected to the switch (A.18) it forces the play mode to loop, ignoring the physical position of the button.

Practice the *Force Loop* with this Technique:
Ratcheting-Like Effect #1

The *Trig/Gate Input* and the *Trig/Gate Button* trig the envelope: you can use either the manual button or an external trig or gate (for details concerning trig and gate

thresholds see the technical specification, p. **Errore. Il segnalibro non è definito.**). If the function is running, any retrig signal recalls the function's rise stage, this means that:

· if the function is already in its *rise* stage, nothing changes;
· if the function is in its *hold* stage, nothing changes;
· if the function is in its *fall* stage, it goes back to rise, starting from the *level* on which it left the fall stage;
· if the function is in its *rest* stage, it starts a new function.

It is useful to know that, being FALISTRI a triangle core generator, and having a wave shaper stage applied after the core, if the retrig happens during fall stage and you are using different wave shapes than linear, some interesting discontinuity will occur.

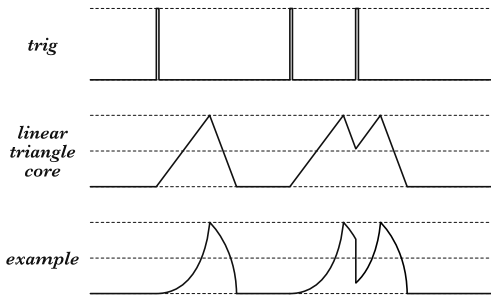


Figure 33: FALISTRI's retrigs with different envelope shapes.

2.3.1 Green Alternative Retrig (On Rest)

The green function generator offers a unique feature which can be activated with a switch on the back of the module. It is up to the user to decide how it should behave:

· *To Rise* (lower position) – same behavior as the yellow one;
· *On Rest* (higher position) – the retrig is effective only when the generator is at its rest stage.

This last feature allows you to use the generator as a formant oscillator, maintaining roughly the spectral content of the function you designed across several notes, and resulting in frequency divisions (/2, /3, /4, /*n*…) of the external frequency used to trigger it, when this "slave" period is longer than the "master" one.

Since this switch is on the back of the module, it is meant to provide a set-and-forget option. However, in case of more than one FALISTRI on the same system, one might actually forget the setting: for this purpose, when the green generator is set *On Rest* and its play mode is *Transient* (see above, §2.3), the trig/gate LED will not light up.

To check whether the green generator is set *On Rest* or *To Rise*, set the generator to *Transient Mode* and push the *Trig/Gate* button (A.2): if the LED lights up, it's *To Rise*; if it doesn't, it's *On Rest*.

Please note that the *Quadrature* mode forces the *To Rise* mode, so the switch is ignored (see below, §2.5.1).

Practice the two behaviors with these *Techniques*:
On Rest/To Rise
Formant/Sync

2.4 Outputs

Each generator provides 5 different outputs: two gates and three CV.

The Control Voltage outputs are:

· Bipolar (A.5, marked with two dots): range ±5V, useful for bipolar LFO or when the function is used as an audio rate oscillator
· Unipolar (A.6, marked with one dot): range 0/+10V, useful for any traditional envelope modulation.
· Attenuverted (A.8, connected to the A.7 attenuverter): ranges from 0V to anything up to +10V or 0V to any down to −10V. It is still unipolar, but with the polarity and amplitude you prefer.

Practice the attenuverters with these Techniques:
Complex Envelopes #2
Kick Drum #1

Gate outputs are, from left to right, *End Of Rise* (EOR) and *End Of Fall* (EOF).

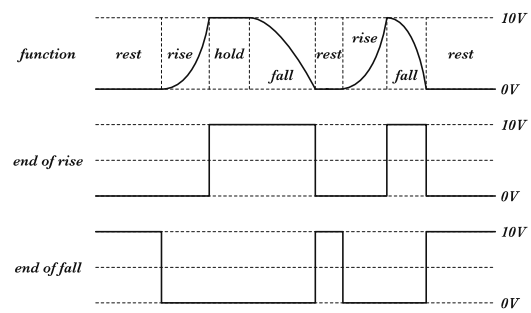


Figure 34: FALISTRI's envelope shapes, EOR, EOF.

2.5 Additional Generator Features

2.5.1 Quadrature

The *Quadrature* switch enables the quadrature mode when moved to the right. *Quadrature* is a peculiar function which allows to chain the stages of both generators to obtain more complex functions. You may want to think as a couple of envelopes whose stages are inter-depending.

Here is the sequence:

- begin of cycle → yellow starts to rise;
- yellow rise ends → green start to rise/yellow in hold;
- green rise ends → yellow start to fall/green in hold;
- yellow fall ends → green start to fall/yellow in rest;
- green fall ends → end of cycle.

To achieve quadrature, move the *Quadrature* switch to the right, and set the green generator in *Hold* mode. You may now achieve three kinds of results, basing on the yellow generator's mode in use:

Yellow in *Loop*: a looping quadrature, which automatically restarts from the yellow *rise* stage when the green ends its *fall*;

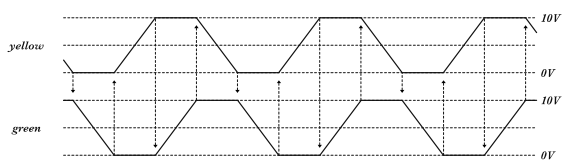


Figure 35: Quadrature mode with the same yellow and green times.

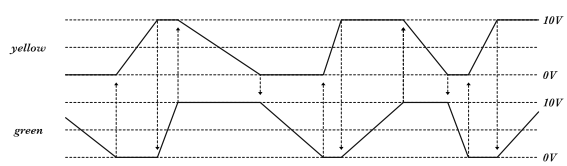


Figure 36: Quadrature mode with different yellow and green times.

Yellow in *Transient*: a single shot quadrature cycle;

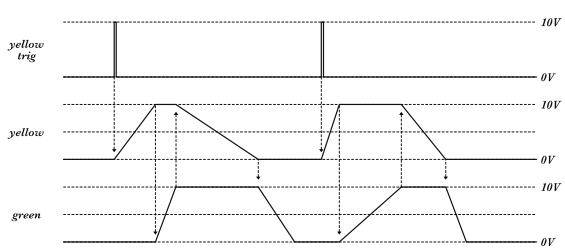


Figure 37: FALISTRI's *Quadrature*, yellow generator in *Transient* mode.

Yellow in *Hold*: a single shot quadrature cycle, whose yellow hold stage depends from both green rise time and the length of the impulse/gate used to trigger the yellow generator.

As described in the *Green Retrig* section (§2.3.1), the quadrature mode deactivates the retrig mode *On Rest* for the green generator.

2.5.2 Max

The *Max* section outputs the higher value between the two attenuverted signals (also called *Analog OR* in some synthesizers). It is particularly useful to combine the yellow and green function in order to achieve more complex envelopes or unconventional wave shapes.

2.5.2.1 ADSR

With *Quadrature* enabled it is possible to obtain envelopes such as ADSR (Attack-Decay-Sustain-Release)/AHDSR (Attack-Hold-Decay-Sustain-Release) by using the *Max output* (E.1) and the *attenuverters* (A.7) to set the relative levels.

The *Attack* is performed by the yellow *Rise*.

The *Hold* time, when needed, is determined by both the green *Rise* time and the yellow *Hold* time (when the yellow generator is in *Hold* mode).

When the yellow *Fall* starts, the green generator enters in *Hold* mode until the yellow envelope reaches its end. The green *Hold* level is set by its *Attenuverter* (A.7).

As long as the yellow *Fall* level is higher than the green *Hold* one, the envelope will be in its *Decay* stage. When the green *Hold* stage level is higher than the yellow *Fall*, the envelope will be in its *Sustain* stage, whose duration is set by the yellow *Fall* time.

When the yellow *Fall* reaches its end, the envelope begins its Release stage, which is set by the green *Fall* time.

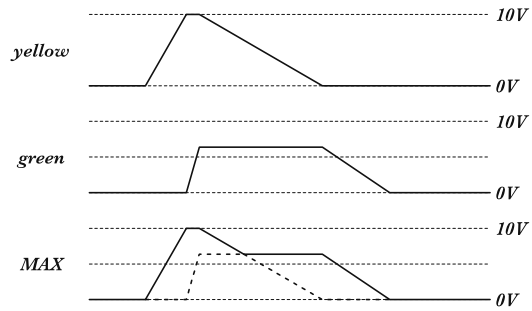


Figure 38: AHDSR envelope with FALISTRI's *Quadrature*.

Practice the *Quadrature* mode and the *Max* output with these *Techniques*:
Quadrature #1
Complex Envelopes #1
Accents #1
Bouncing Ball

3 FUNCTION PROCESSORS

3.1 Dual Cascaded Frequency Divider

Also known as *flip/flop* or *sub-octave processor*, each of these processors changes its logic state every time a low to high transition is detected.

The top left jack sockets (D.1, D.2) are inputs, the bottom right ones (D.3, D.4) are outputs. The first circuit is the top one, the second circuit is the bottom one.

The two sections are semi-normalled in order to achieve 1/2 and 1/4 outputs from a single input. Output *Range* switches (D.5, D.6) are useful to obtain unipolar (0V/+10V) when the switch is high or bipolar signals (±5V) when the switch is low.

When an audio signal is used as input, it works as a sub-octave generator (a square or pulse wave is preferred as carrier). When a clock is used it works as a clock divider. Bear in mind that if the input is a trig, the output is always a gate, since the flip flop retains its state (high or low) until a new rising edge is detected at its input.

Random clocks from SAPÈL create interesting variable-length sequences of gates.

The following example uses a clock as input, obtaining half and quarter of that clock frequency.

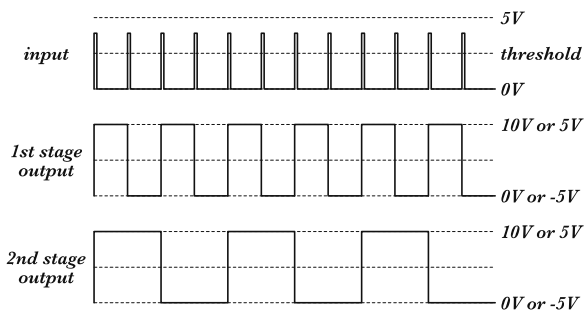


Figure 39: FALISTRI's *DCFD* as a double clock divider.

The following example uses a random clock stream as input.

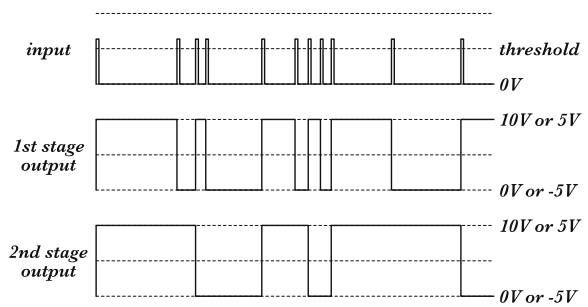


Figure 40: FALISTRI's *DCFD* as a random flip-flop.

This last example uses a square-ish wave as audio rate source, obtaining −1 and −2 octave as output.

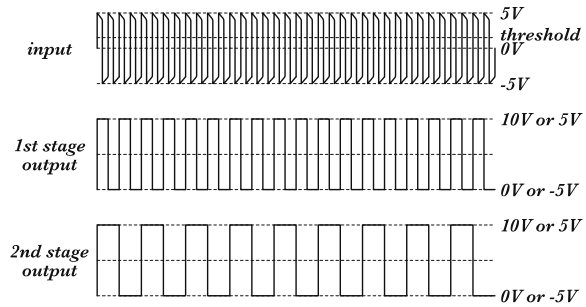


Figure 41: FALISTRI's *DCFD* as a sub-octave generator.

Practice the *DCFD* with these Techniques:
Ratcheting-Like Effect #1
Octave Blends

3.2 Four-Quadrant Multiplier

The *Four-Quadrant Multiplier* (4QM) is a circuit that multiplies two signals by one each other. It is similar to a VCA, with the main difference that it welcomes bipolar signals on both inputs, allowing any possible combination of positive and negative polarities, which can be thought of as the four quadrants (hence its name) of a two-dimensional Cartesian plane. It can be thought of as a ring modulator, a balanced modulator, or a through-zero VCA.

With an exceptional linearity, and a bandwidth from DC to more than 20KHz, the result of this operation is completely up to the sources in use.

The two inlets are the jack sockets on the left: the first from top (B.1) is semi-normalled to the yellow unipolar output; the second (B.2) is semi-normalled to the green bipolar one: in this way you can use the yellow as an envelope and the green as an audio source, without patching any cable.

The *Level* knob (B.3) sets the amount of the two signals, affecting the *Output* as a whole (B.4).

The 4 led matrix shows which of the quadrant is currently in use: the *Input 1* (B.1) moves the lights vertically (negative bottom, positive top), and the *Input 2* (B.2) moves the lights horizontally (negative left, positive right.

3.2.1 Amplitude Modulation & Ring Modulation (2 vs 4 quadrants)

When one of the signals is bipolar and the other is unipolar, the *4QM* works like an actual linear VCA: this is the default behavior. This means that if, for example, the green generator is working as an oscillator, the yellow one controls its amplitude.

This configuration allows you to use FALISTRI as a simple, yet effective synth voice. Practice this Technique:
Synth Voice

If the yellow generator too is brought to audio rate, FALISTRI starts to perform amplitude modulation (AM), also called unbalanced modulation because one of the two signals is positive-only and only two of the four quadrants of our hypothetical Cartesian plane are in use.

The result of the amplitude modulation is a signal which retains the carrier frequency with additional frequencies called 'sidebands': in case both signals are pure sine waves, these sidebands will be two, that is the sum and the difference of the Carrier and Modulator's frequencies, whose amplitude depends on the amount of modulation, but in any case will not exceed half of the Carrier's amplitude. In case the Modulator's frequency is higher than the Carrier's, and their difference results in "negative" frequency, they will be heard with inverted phase.

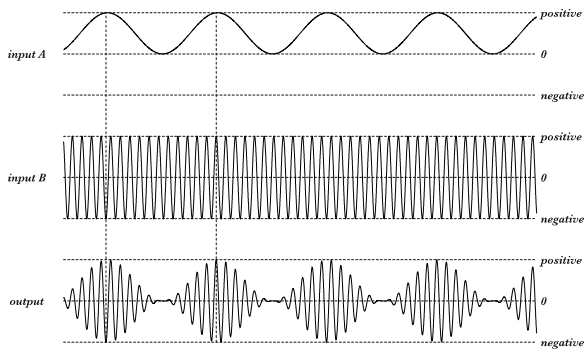


Figure 42: Amplitude modulation (AM).

In case both signals are bipolar, FALISTRI starts to perform the so-called Ring Modulation (RM), or 'balanced modulation' because it uses all of the four quadrants (i.e. the combinations of positive and negative polarities).

The result of RM is quite similar to the one of AM, with the major distinction that in RM the carrier frequency is suppressed, being the sidebands the only audible output. Their amplitude, in this case, is equal to the Carrier's one.

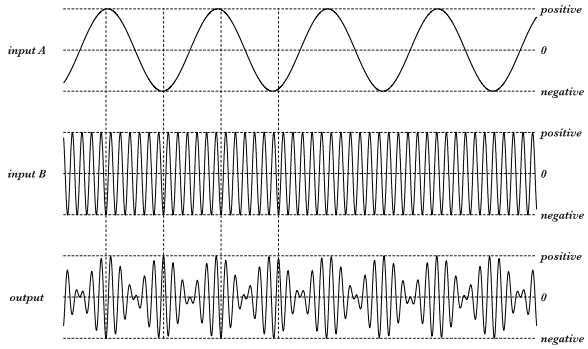


Figure 43: Ring modulation (RM).

Practice AM and RM with this Technique:
VCA/AM/RM

3.2.2 Trimming

The front panel features a trimmer in the 4QM area. This may be useful to set the zero level when used with a unipolar envelope that falls at zero, reducing possible DC offset that would cause a lower attenuation. By default, it is trimmed with the yellow unipolar signal.

3.3 Linear Slew Limiter

Also known as *portamento* or *glide* when using a quantized CV, this processor comes into play when any voltage smoothing is needed. This section smoothens any kind of voltage transition with independent controls over the rising and falling edges of the incoming signal.

The pot on the left (C.2) sets the slew of the rising voltages patched to its input (C.1), while the pot on the right (C.3) of the falling ones. The result is routed to the *Output* (C.4) The two LEDs are also helpful to display what is happening under the hood.

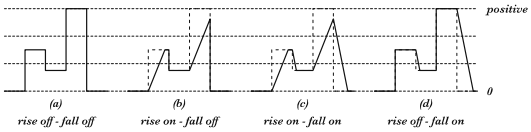


Figure 44: Linear slew limiting.

Practice this tool with the following Technique:
Pitch Smoothing

4 TECHNICAL DATA

4.1 SPECIFICATIONS

Parameter	Details		Min	Typ	Max	Unit
Current Draw		+12V			170	mA
		-12V			170	
Size				18		HP
CV input impedance				> 90		KΩ
Clock Input	Input Trigger Threshold	Amplitude		> 1.5		V
		Maximum Frequency (at 1.5V)		> 6		KHz
		Minimum Pulse Period		< 80		µs
		Impedance		> 90		KΩ
Loop input	Input Trigger Threshold	Amplitude		> 3		V
		Maximum Frequency (at 3V)		> 6		KHz
		Minimum Pulse Period		< 80		µs
		impedance		> 50		KΩ
Times & Frequencies	Single stage (Rise or Fall) with Long Times	from		> 0.9		ms
		to	> 8	> 10		s
	Complete Cycle with Long Times	from		< 1.9		ms
				> 520		Hz
		to	> 16	> 20		s
			<0.625	< 0.05		Hz
	Complete Cycle with Short Times	from		< 0.16		ms
				> 6.1		KHz
		to	> 1	> 1.5		s
			< 1Hz	< 0.67		Hz
Generators Voltages Output		Unipolar Amplitude		> 10		V
		Bipolar Amplitude		> ±5		V
		Unipolar Offset (in Rest)			< 35	mV
		Impedance		> 100		Ω
Generators V/oct Tracking		audible range		< ±5		¢
MAX output		Tolerance				%
		Impedance		> 100		Ω
Generators Gate Output		Amplitude		> 10		V
		Offset	-20		20	mV
		EOF min positive pulse period	1			ms
		Impedance		> 220		Ω
Cascaded Flip-Flop(s)	Input Trigger Threshold	Amplitude		> 3		V
		Maximum Frequency (at 3V)		40		KHz
		Minimum Pulse Period		<10		µs
	Output	Unipolar Amplitude (high)	>9	~9.5		V
		Bipolar Amplitude	>±4.5			
		Offset		<50	<100	mV
	Impedance	Input		>90		KΩ
		Output		>220		Ω
Four Quadrant Multiplier	Dynamic Range (1)(2)		75	78		dB
	Residual THD+N (1)			0,015	0,02	%
	Maximum Allowable Gain (1)(3)		6	6,5	7	dB
	High Frequency Roll Off	-3dB		36		KHz
	Impedance	Input		>90		KΩ
		Output		>100		Ω
Slew Limiter	Slew Rate	Minimum	1		1,25	V/ms
		Maximum	0,5	0,6	0,7	V/s
	Impedance	Input		>28		KΩ
		Output		>100		Ω

(1) Using a DC on one input and a sine wave at 2.4KHz at 3V rms on the other. Pot fully clockwise.
(2) Filtered with 30KHz LPF and 400 Hz HPF.
(3) 10Hz÷80KHz

4.2 REVISIONS

Starting from lot no. 201001 we rearranged the board layout to facilitate the calibration procedure in the lab. The functions of the modules and its tech specs have not been changed.

(Refer to firmware v. 152)

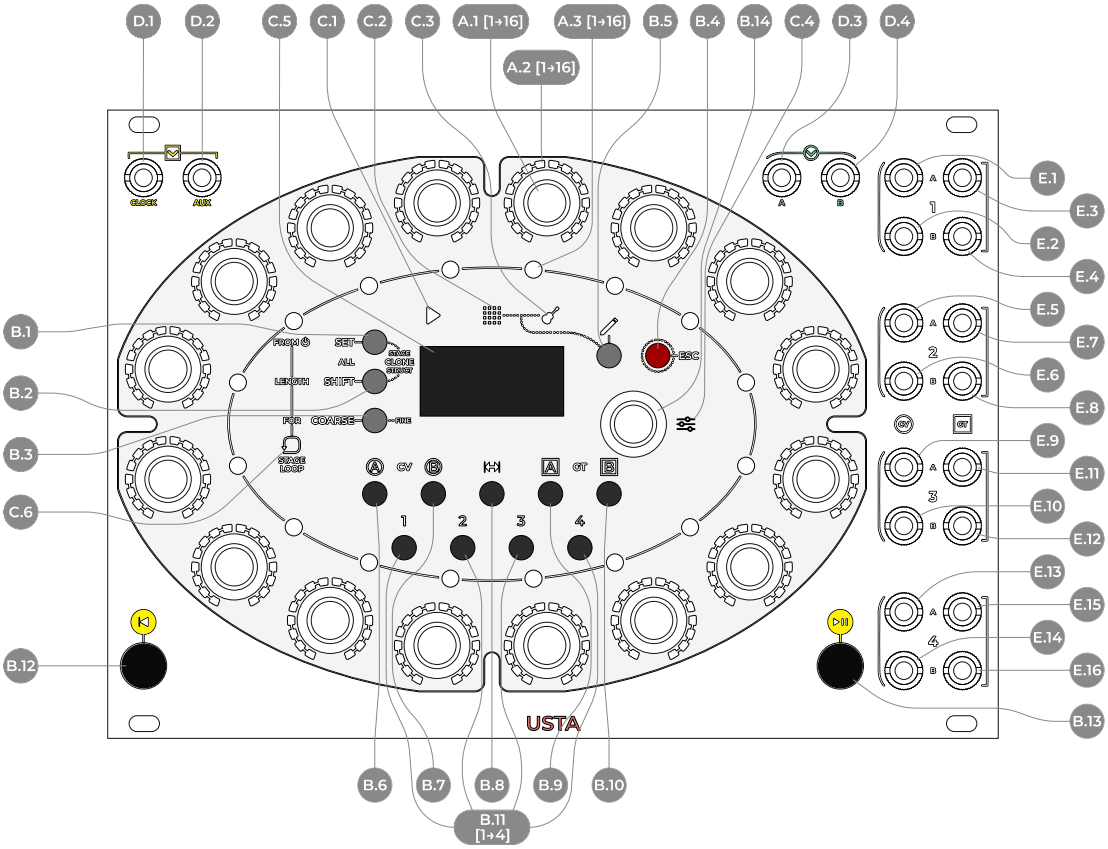


Figure 45: USTA interface.

A	**Stages**	**B.12**	Reset	**E.1**	Track 1 – CV A
A.1	Stage Encoders [1→16]	**B.13**	Play	**E.2**	Track 1 – CV B
A.2	Stage Arc	**B.14**	Navigation Encoder	**E.3**	Track 1 – Gate A
A.3	Stage RGB LEDs	**C**	**Other Visual References**	**E.4**	Track 1 – Gate B
B	**Buttons and LEDs**	**C.1**	Play LED	**E.5**	Track 2 – CV A
B.1	Set All	**C.2**	Pattern LED	**E.6**	Track 2 – CV B
B.2	Shift All	**C.3**	Song LED	**E.7**	Track 2 – Gate A
B.3	Coarse	**C.4**	Menu LED	**E.8**	Track 2 – Gate B
B.4	Esc	**C.5**	7-Segment OLED Display	**E.9**	Track 3 – CV A
B.5	Pencil Button and LED	**C.6**	Stage Loop LED	**E.10**	Track 3 – CV B
B.6	CV A button and LED	**D**	**Inputs**	**E.11**	Track 3 – Gate A
B.7	CV B button and LED	**D.1**	Clock Input	**E.12**	Track 3 – Gate B
B.8	Length button and LED	**D.2**	Auxiliary Gate Input	**E.13**	Track 4 – CV A
B.9	Gate A button and LED	**D.3**	CV A Input	**E.14**	Track 4 – CV B
B.10	Gate B button and LED	**D.4**	CV B Input	**E.15**	Track 4 – Gate A
B.11	Track buttons and LEDs [1→4]	**E**	**Outputs**	**E.16**	Track 4 – Gate B

1 QUICK START

Once you have installed USTA and powered up your system, you will see a fast boot screen with the Frap Tools logo on the OLED Display (C.5), then a screen with various details concerning the project you are about to create. *Track 1* (B.11) and its *CV A* (B.6) are already selected and ready to be edited.

Start rotating the 16 encoders (A.1) and you will see that the LEDs surrounding them (A.2) are progressively lighting up, and at the same time a changing note name appears in the lower-left corner of the Display, indicating the pitch of the selected stage of the sequence. This is the content of the sequence, which will be outputted by the *CV A* jack port of the first track (E.1).

Patch it to the 1V/oct input of your oscillator and then push the *Play* button (B.13) to hear the sequence.

Now you may need gate signals in order to separate the notes and add dynamics to your sequence. Push the Gate A button (B.9) to access the *Gate Channel* and rotate the *Stage Encoder* (A.1) as before.

You are now defining the individual gate-high time for each of the 16 notes you entered. Patch the *Gate A* output of Track 1 (E.3) e.g. to an envelope controlling the VCA, or the filter.

What you just obtained is a sequence of 16 notes with equal duration. Feel like something is missing? It actually is. In order to learn how to work with full polyphony, have faster or slower sequences at the same time, change the length of the notes, make a single note vary every time it is played, but still in the key you chose, have some random ratcheting on a given stage, loop a section within a song structure, and many other things, please take a deep breath and keep reading.

2 PHILOSOPHY AND DESIGN

USTA is a 4×4 tracks, variable-stage-length sequencer for voltages and gates.

Variable stage length means that every single stage duration can be individually set in relation to the clock, instead of being constrained to a one-to-one ratio (i.e. one step per every clock impulse).

4×4 tracks means that every *stage* can store and generate up to four separate voltages (two CV and two gates) and that up to four independent sequences can be arranged as simultaneous tracks.

USTA's user interface is based on a straightforward 16-stage circular design with an array of multi-functional encoders with push switch and buttons. This allows users to easily navigate within the sequencer's flexible software architecture and to quickly display and edit relationships between the different stages.

Those main features are what makes USTA the Eurorack *Voltage Score*: its flexibility on the time, rhythm and harmony domain, its large set of independent outputs and its performance-oriented interface allow the user to achieve nearly any musical result, from fine Baroque polyphony to the most extreme polyrhythmic noise live performance.

2.1 ARCHITECTURE

USTA's architecture is designed to provide a wide set of compositional tools without sacrificing the user-friendliness.

The smallest "brick" of the sequencer is the *stage*, which contains two separate control voltage values, two separate gate values and all the behavioral information necessaryto generate four synchronous musical events, including pitch, volume, timbre, repetition, probability…

A sequence of up to 16 stages (one per encoder) forms a *pattern*, in the same way that a series of individual musical events forms a melody.

Up to 32 patterns are contained in a *track*, which can be compared to a musical piece for solo instrument. Beside all the melodic content stored in a pattern, a track contains also the performance information, including: which pattern is currently playing, which external control is used as modulator, and so on. Four tracks are simultaneously available in USTA, and they can have different tempos, number of stages and patterns.

Just like any musical piece, the track can be either improvised, when the musician decides all these performance details on the go, or carefully composed first, and then played back: in order to accomplish this last task, USTA allows an alternative playing mode called *song*, in which all the patterns can be arranged in a pre-determined set.

This information can be independently set per each of USTA's four tracks. All the four tracks along with their data and information are contained in a *project*, which is the general operational framework of the sequencer. A project can be stored on the included micro SD and recalled whenever is needed (one project at a time). On the first startup an empty project is created (called NONAME), then at every other startup USTA automatically will recall the last project opened how it was saved. Projects and trimming settings can be backed up on your computer from the micro SD.

2.2 TEMPO MANAGEMENT

Conventional step sequencers use the reference clock to jump from one step to the other. This means that the step duration, which is the time from one step to the other, corresponds to the period between consecutive clock impulses.

USTA, on the other hand, allows every stage to have a different length that can be set in relation to the reference clock (and this is one reason why the term *stage* is preferred over *step*).

Such relation is achieved through two deeply connected concepts, *time units* and *time ratio*. The *unit* is the shortest possible duration of a given stage, while the *ratio* defines the unit length in respect of the clock tempo.

It is possible to have 16 different time ratios: 8 are clock divisions, thus providing units that are longer than the clock impulses; 7 are clock multiplications, thus providing units that are shorter than the clock impulses; one is the exact clock period, thus providing units that are as long as the clock impulses.

The following chart displays the clock to unit ratios available: on the left is the number of clocks, on the right, the number of units.

24:1	Clock division (n clock impulses per 1 unit)
8:1	
7:1	
6:1	
5:1	
4:1	
3:1	
2:1	
1:1	1 clock impulse per 1 unit
1:2	Clock multiplication (1 clock impulse per n units)
1:3	
1:4	
1:5	
1:6	
1:7	
1:8	

Table 3: Clock-to-unit ratio

Each *stage length* can measure from 0 to 16 units, where 0 means that the stage has no length in the time domain, therefore it will skip.

The combination of time ratio and units allows extremely flexible sequences of stages. For example, with the same clock, it is possible to achieve fast sequences if the time ratio is set to 1:4 and the stage duration is set to 1 unit, or slow sequences of whole notes if the time ratio is set to 2:1 and the stage duration is set to 4 units. It is also possible to have extremely slow sequences by cranking up the stage length to 16 units and the ratio to 24:1…

Another benefit of USTA's architecture is the efficient use of stages to create melodies: in the image below, the melody (a) needs 16 steps and four gates to be performed on a standard step sequencer with a regular clock (b), while on USTA it takes only four stages, one per each musical event (or "note"), each one with its relative duration and gate (c).

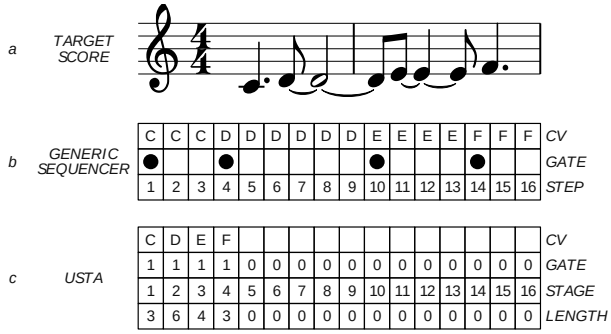


Figure 46: Step versus stage sequencing.

3 BASIC EDITING AND VISUAL FEEDBACK

The interaction between the musician and USTA happens both through the navigation menu and more "manual" operations such as button and encoder combinations. Likewise, the visual feedback combines the color-coded LEDs and the information provided by the default screen on the display (C.5), called *Dashboard*.

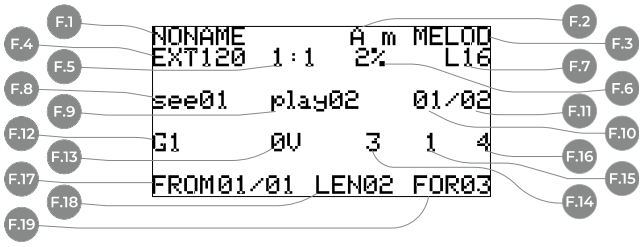


Figure 47: USTA's *Dashboard*.

F **Dashboard**
F.1 Project Name
F.2 Root Note
F.3 Scale
F.4 Clock Source and BPM/PPM
F.5 Time Ratio
F.6 Swing
F.7 Total pattern length
F.8 Selected pattern
F.9 Playing pattern
F.10 First pattern
F.11 Last pattern
F.12 CV A value
F.13 CV B value
F.14 Length Value
F.15 Gate A value
F.16 Gate B value
F.17 Stage Loop From
F.18 Stage Loop Length
F.19 Stage Loop For

As a rule of thumb, all the data regarding the general behavior of USTA (track settings) are accessed through the *Project Menu* and *Track Menu* (see below, §§3.1-3.2), while all the data concerning the very musical content, such as individual stage data, are set via dedicated encoders and buttons (§3.3 onwards).

3.1 EDITING PROJECTS – PROJECT MENU

On its first boot USTA automatically creates an empty project called `NONAME`, which is ready to be edited. All the editing is stored in a volatile memory: to avoid data loss, it is possible to save the project into an SD card. To perform these tasks, push and hold the navigation encoder (B.14) for three seconds until the *Menu LED* (C.4) lights up red: this will open the project menu.

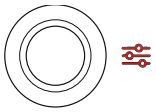


Figure 48: Project Menu LED.

Once there, rotate the same encoder to navigate through the menu items, and push it to select the desired one.

It is possible to perform five operations concerning projects:

`New...`	create a new project, which will be called NONAME
`Save as...`	save the current project with a new name
`Load...`	load a previously saved project;
`Save`	save the current project
`Delete...`	delete a previously saved project
`Rename...`	change the name of a previously saved project.

When performing tasks such as `Load...`, `Delete...` or `Rename...`, a new window will open in which all the projects are listed: rotate the navigation encoder to select the desired project and push it to perform the desired task.

When saving the project as a new one, or when renaming an existing one, a new window will open in order to let you type in the new name. You will notice an angle bracket after the last character of the name (<): this is the *delete* symbol, and when it is present you may press the encoder to delete a letter or number.

In order to add a new character, rotate the encoder and skim through the alphabetical series. Push the encoder again to confirm and move on to the next character. Once the new letter is confirmed, the *delete* symbol will appear again and it will be possible to delete the last entered character.

Once the name is set, press the *Play* button (B.13) to save it and the Reset to cancel the operation.

These operations are not designed for live performance; still, they can be easily performed while USTA is playing, without major issues. In case a project is saved while USTA is playing, the machine temporarily freezes on the current step, then gets back to play. When loading a project while playing, USTA does not stop, but all the user interface elements (LEDs, OLED screen, RGB LEDs and Stage Arc) are momentarily disabled for half a second. During these operations, USTA might slightly slow down or even miss a clock impulse when using an external clock, with direct consequences on the Clock-to-Unit Ratio calculation. Please note that you cannot rename a freshly created project unless you save it first.

A high number of projects can be stored into USTA's microSD, but only the first 128 projects in alphabetical order will be used/recalled.

Once you are done editing the project, press Esc to return to the dashboard: the name of the selected project will then be shown in the top section (F.1).

If you happen to select `Save` instead of `Save as`... USTA will automatically save your current project as `NONAME`; however, this procedure is not recommended since any other project accidentally saved as `NONAME` will overwrite this last one.

3.2 Editing Tracks – Track Menu

Once a new project is created, you can start to compose or improvise. USTA allows you to edit one track at a time. Track 1 is selected by default: to select another track, press one of the four *Track Buttons* (B.11).

Once the desired track is selected, the LED above its button will light up green. At this point, USTA's interface displays the information relative to the selected track and it is possible to play or edit it.

3.2.1 Clock Settings

To set the clock, push once the *Navigation Encoder* (B.14) to access the track menu: the *Menu LED* (C.4) will light up green.

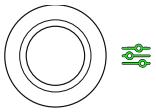


Figure 49: Track Menu LED

First of all, it is possible to choose whether your clock will follow an internal or external clock (see below §7.1).

If you choose to have the selected track controlled by the internal clock, you shall then define the internal tempo (expressed in BPM) by editing the `Int BPM` parameter.

Whether you choose to use an internal or external clock, you can select the time ratio on which calculates the step unit, through the `Ratio` parameter.

If you want to apply the same *Track Menu* settings to all the tracks, hold the *Set All* button (B.1) before pushing the navigation encoder to save the setting (see below §4.1).

Once you finished, press *Esc* (B.4) to exit the track menu and return to the dashboard: the information concerning BPM and time ratio is now displayed in the second row, right below the project name (F.4, F.5).

If you chose to use the internal clock, the tempo value is expressed in BPM, after the abbreviation `INT`; if you chose to use an external clock, patch it to the *Clock Input* (D.1): USTA will calculate its Pulses Per Minutes, which will be displayed on the *Dashboard* after the abbreviation `EXT`.

Practice the clock settings with this Technique: Clock and Ratio

3.3 Editing, Playing and Looping Patterns

Once the time dimension has been set, it is time to define the content of your track, that is to say, the *patterns*. 32 patterns are available per track, which can be edited one at a time. Please note that the patterns are not "created": instead, they are always present in each track, with all the values set to 0, and ready to be edited. The default play mode of USTA is *Pattern Mode*, where a loop of

pattern is played in numeric order. (It is possible to change this order through the *Song Mode*, on which see §5.3) When a new project is created, only the first pattern is looping in *Pattern Mode*: let's focus on this one for now.

Pattern Mode is displayed by the *Pattern LED* (C.2) when it's lit up green.

USTA allows you to perform two main kinds of task with patterns: editing them and playing them back. These two operations can be performed separately (in a more "compositional" approach) or simultaneously (in an improvised performance), but always require two different operating modes, respectively called *Edit* and *Performance*.

These modes are accessed through the *Pencil* button (B.5), which switches from one to the other, and their status is displayed by the pencil-shaped LED: red when in *Edit* mode, green when in *Performance*.

When the pencil LED is red, USTA is in edit mode, which is also the default setting once a project is created and a track is selected. In this mode, you can modify whichever parameter you want.

As said above, it is also possible to play the sequence while editing: the visual feedback for the current stage playing is provided by a cyan stage RGB LED, called playhead. The playhead is displayed only when the selected pattern is playing. For example, when editing pattern 2, if pattern 2 is playing, the playhead is displayed, but if pattern 1 is playing, no playhead will be displayed.

In performance mode – the pencil RGB LED is green – the displayed pattern is always the one that is playing, according to the track: in this mode, it is no longer possible to edit patterns, but other operations are possible (see below §5.4).

The third row of the *Dashboard* (Figure 47, p. 48) displays all the information concerning the patterns that are currently being edited or played.

`see` (F.8) indicates the pattern that is currently selected for editing purposes. In order to select a pattern, make sure to be in *Edit Mode*, then rotate the *Navigation Encoder* (B.14). You will see this number changing accordingly.

`play` (F.9) indicates the pattern that is currently being played.

The two numbers separated by a slash indicate the beginning (F.10) and the ending point (F.11) of your pattern structure. By default, they are set to `01/01`, which means that the pattern loop will be limited to the first pattern only. It is possible to edit any other pattern while pattern 1 is playing, however, the result of such operation will not be heard unless the newly edited patterns are included within the pattern loop.

To change the extension of the pattern loop, enter *Performance Mode*, then hold the *Set All* button (B.1) and rotate the *Navigation Encoder* (B.14) to set the first pattern of your looping structure, or hold the *Shift All* button (B.2) while rotating the *Navigation Encoder* to set the last one. From now on, USTA will loop all the patterns contained within those two, including them.

Please note that the last pattern needs to be higher than the first pattern (or the same), because in Pattern Mode the playhead moves from one pattern to the other in numeric order. To obtain different pattern orders, please refer to the *Song Mode* (§5.3).

Another way of editing the pattern loop is through a button combination. When USTA is in Performance mode, every editing option is disabled: the 16 *Stage Encoders* (A.1) are now associated with the first 16 patterns, and while holding the *Shift All* button (B.2) they are associated with the last 16 patterns (or from the 17th to 32nd). They can be pushed to manually recall a given pattern (on which see *Pattern Recall*, §5.4), but they can also be used to define the pattern loop on the fly. Push and hold the *Set All* button (B.1) and then push the encoders corresponding to the first and last patterns to set the new ends of the pattern loop: remember to hold also *Shift All* (B.2) to set patterns no. 17-32.

If you want to apply the same pattern loop settings (first and last pattern) to all the tracks at the same time, double click *Shift All* when in *Performance Mode*.

Practice the Pattern Loop with these Techniques:
Edit & Performance Mode
Pattern Loop

3.4 Editing Stages

In order to modify the content of your pattern, you need to edit its stages. Each stage can have its own *length*, expressed in terms of *units*.

Furthermore, each stage can store information for two separate control voltages and two gates, which form each stage's *channels*.

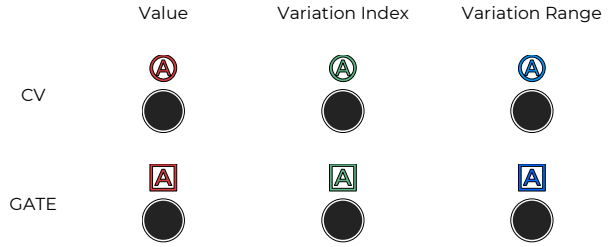


Table 4: Channel LEDs

To access and modify these data, it is required to select the desired channel through the four *Channel Buttons* (B.6, B.7, B.9, B.10) or the *Length Button* (B.8) and then modify its value with the respective stage encoder (A.1). Each encoder corresponds to one and only one stage per pattern, numbered clockwise from the topmost one on the right.

Both CV and Gate channels have three different *layers* of data: *Value*, *Variation Index* and *Variation Range*. By

pressing a channel button (B.6, B.7, B.9, B.10) multiple times, the user cycles through its layers. The currently selected layer is indicated by the LED color above the channel button (see Table 4).

An arc of 16 yellow LEDs, the *Stage Arc* (A.2), provides visual feedback for the current value selected or edited per each stage. Furthermore, an RGB LED per each stage (A.3) provide information about the *stage color*, which defines the way a CV or gate is played (slide, freeze, ratchet, tie... see below §§3.4.4-3.4.6). This color is temporarily overwritten by the cyan *playhead* when that stage is played.

All the data of the stage which is currently being edited are also displayed in real time in the fourth line from the top of the dashboard.

The five numbers are in the same order of the channel buttons: from left to right, they display the value of the last edited stage's *CV A* (F.12), *CV B* (F.13), *Length* (F.14), *Gate A* (F.15) and *Gate B* (F.16).

When entering *Performance Mode* (green *Pencil* LED, B.5), this row of the dashboard can be set to display, in real time, the values of the stage that is currently being played (see below, §8.4).

The four outputs per track are grouped on the right side of the front panel (E.1 to E.16). The number in each group refers to the track and the letters to the individual CV or Gate channels. As a rule of thumb, the symbols with a letter in a circle refer to CVs, while the ones with a letter in a square refer to gates.

Each pattern by default has a sequence of 16 stages with equal length of one unit, whose speed depends on the clock and the time ratio previously selected. First of all, let's edit the individual stage length.

3.4.1 Length

The *Length* section determines the duration of each stage in terms of *units* (see above), and it is accessed by pressing the *Length* button (B.8).

By default, the length of each stage is 1 unit. In order to increase it, turn the respective encoder (A.1) clockwise. Each step of the encoder increases the stage duration by 1 unit, up to a maximum value of 16: the *Stage Arc* (A.2) will light up accordingly, displaying the exact number of units per stage.

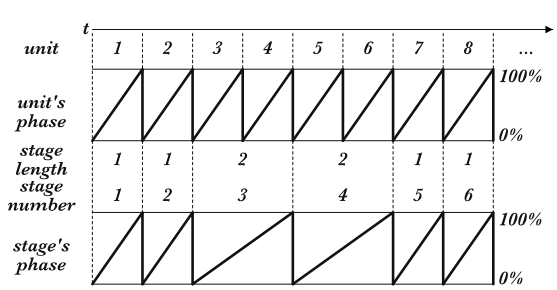


Figure 50: *Stage Length* and stage phase.

It is possible to turn the encoder counterclockwise until no LEDs is lit: in this case, the stage will have length 0, so it will be skipped during the playback.

When a given stage's length is being edited, the unit number is displayed also by the third digit from the left of the fourth line from the top of the dashboard (F.14).

The total length of a pattern is displayed at the right of the second row from the top, beside the letter L: this value may come in handy when you want to edit the length of two stages without changing the overall pattern length, as described in the following section.

Practice these basic concepts with the following Techniques:
CV, Gate, Length
Stage Vs. Gate

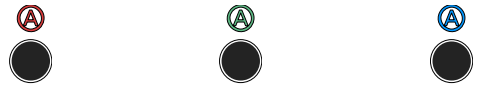
3.4.2 Maintain Pattern Length on Variation

While performing it may be useful to hold the length of your pattern fixed while editing stage lengths. Adding or subtracting units can unintentionally shift the pattern's rhythm out of alignment. To reduce this chance, hold down the *Coarse* button (B.3) while changing the stage length. This causes a neighboring stage length to receive a compensating adjustment.

For example, assuming your pattern has only the first 4 stages with length 1 and the others with length 0, rotating the stage 2 encoder clockwise will give stage 2 a 2-unit length for a pattern length of 5 units. Rotating the encoder counterclockwise gives stage 2 a 0-length stage, resulting in a 3-unit length pattern. If these steps are performed while the *Coarse* button is held down, the clockwise rotation will increase the stage 2 length to 2, but the 3rd stage will be reduced to length 0 to compensate. Likewise, a counterclockwise rotation will decrease the stage 2 length to 0, while the 3rd stage will be increased to length 2. In both cases the pattern length of 4 units is retained.

3.4.3 CV Layers

The CV channels' parameters can be modified through three different *layers*, which are accessed by pressing the respective channel button (B.6, B.7) multiple times.



Sets the stage CV value as notes or volts.	Sets the chance to variate the defined value for the stage.	Sets the bipolar range for the variation as notes or volts.

Table 5: CV Layers Comparison.

The first layer is marked by a red LED and determines the CV value (such as the pitch), while the second and

third layers (respectively marked by green and blue LEDs) manage possible variations of the red layer's value.

3.4.3.1 Red CV Layer: Value

Figure 51: CV A's *Value* layer is selected.

The red layer is displayed by default once a channel is selected. It contains the data concerning the stage *value* (i.e. the actual voltage that will be output by USTA).

Such values can be quantized (as in 12 semitones) or raw (i.e., continuous voltages with down to 1mV of resolution). The default setting is *Pitch* for *CV A*, which can be used for generating melodic lines, and *Raw* for *CV B*, which can thus be used to add dynamics and to modulate other "musical" parameters such as filters, VCAs, timbre… It is possible to change these settings and have both CVs in raw or quantized mode (see below §8.1).

In order to edit this value, select the desired channel (*CV A* or *CV B*) through its button (B.6, B.7), then rotate the encoder corresponding to the step you want to edit (A.1). Each time you edit a stage value, the fourth row from the top of the dashboard updates showing the *CV A*, *CV B*, *Length*, *Gate A*, and *Gate B* for that stage on that pattern of that track: this is helpful to understand relationships of the same stage across channels.

Pitch CV channels display notes using the standard notation letters from A to G followed by the octave number, while *Raw* CV channels display Volts. The octave number can be set to change on A or C through the reference note (see below §9.4).

In *Pitch* mode, the *Stage Arc* (A.2) will also precisely display the selected note. 12 LEDs from left to right display the note value in half steps: none for C, one for C♯/D♭ and so on until eleven lit LEDs stand for B. The note references apply when the project reference note is C. To change it to A (see below §9.4). The four remaining LEDs will display the octave, from right to left. For a precise chart of this visualization system see *LED Pitch Tables*, §9.6.

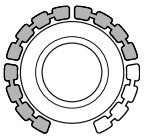


Figure 52: The grey LEDs display the semitones.

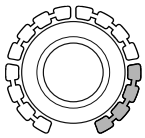


Figure 53: The grey LEDs display the octaves.

By default, USTA works with the 12-semitone equal temperament, which divides the octave into 12 equally spaced intervals. It is possible, however, to change this setting and work with other more complex octave divisions such as 15, 19, 22 or 24 intervals, on which see below §9.2.

For raw channels, such as channel *CV B* by default, the encoder increases the stage CV value by steps of 0.05V or 50mV when turned clockwise. In this mode, coarse and fine edit is possible: with the Coarse button held down, the encoder steps are of 0.5V (500mV), while with the fine (Esc held down and Coarse held down) steps are of 0.001V (1mV). Visual feedback of the value is displayed on the 16 yellow LED of each Stage Arc, scaling the 10V of range to 0/16 LED: in other words, each LED shows 0.625V.

Practice the *Raw* voltages with this Technique: Raw CV

3.4.3.2 Green CV Layer: Variation Index

Figure 54: CV A's *Variation Index* layer is selected.

The green layer is accessed by pressing the channel button (B.6 or B.7) a second time: it controls the probability that USTA will shift the red value up or down in a given bipolar range (see below, Blue CV Layer).

By default, the stage values are at 0, with the *Stage Arc* (A.2) completely off: this means that there is no chance that the note (or voltage) will change, so USTA will stick to the value assigned in the red layer.

By turning the *Stage Encoders* (A.1) clockwise, you increase the chance that the note will be replaced with another one (*Variation Index*), picked by a pool of values whose range is defined by the blue layer (*Variation Range*, see the following chapter). In this mode, USTA tosses a coin at every stage with an *Index* bigger than 0 and decides whether the stage will pick the *Value* defined in the red layer or shift it.

When 8 LEDs out of 16 of the *Stage Arc* are lit up, the chances are 50-50: the probability that the defined value will be played or not are the same. When all the 16 LEDs are lit, it is certain that another note or voltage will be evaluated.

It does not mean, however, that the *Value* defined in the red layer will not play at all, since it is still within the *Variation Range* defined by the blue layer, so it is possible that it might be picked by the coin toss.

3.4.3.3 Blue CV Layer: Variation Range

Figure 55: CV A's *Variation Range* layer is selected.

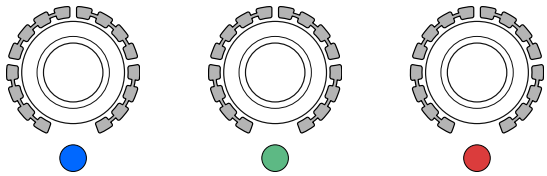
The blue layer is accessed by pressing the channel button (B.6 or B.7) a third time: it determines the range of values which the variation index will refer to (the *Variation Range*, i.e., the values that may or may not be "picked" by USTA instead of the one defined in the red layer).

The blue layer lets you choose the bipolar range of values that could be selected by USTA after tossing the coin. Such values are expressed in semitones for the *Pitch* channels and millivolts for the *Raw* ones. The range increments by steps of ±2 semitones each for the quantized channels (up to ±32 semitones) and in steps of ±314 mV each for the raw channels (up to ±5.024 V).

While some extreme settings of the green and blue layer might translate into a behavior comparable to the one of a random voltage generator such as SAPÈL, their overall approach is completely different, if not the opposite. SAPÈL is a true random module whose voltages are sampled using analog noise as a source, therefore there is no chance that a sequence of values will repeat itself over time. USTA, on the other hand, uses a digital "coin toss" algorithm to randomly pick a value within a given range defined by the user, whose result totally depends on the *Value* chosen as default. In other words, while SAPÈL shapes its stream of random voltages by "subtraction", i.e. through sample-and-hold, quantization, probability distribution, USTA progressively "expands" the range of values according to the musician's instructions.

3.4.4 CV Stage Colors

Each stage can be played in three different ways called *Stage Colors*, which can be accessed by pushing the *Stage Encoder* (A.1) multiple times. In order to better understand the relationship between *Stage Colors* and the *Layers* described in the previous chapters, one could say that *Layers* affect the *content*, i.e. the values that the stage will play, while *Colors* affects the *form*, i.e. the way in which such values will be played.



The defined voltage is generated for the whole stage length.	A ramp is generated for the whole stage length from the previous stage's value towards the current.	No voltage is generated: the last previous stage's value is held for the whole current stage length.

Table 6: CV Colors comparison.

The first CV *Stage Color*, called *Flat*, is selected by default once a channel (*CV A* or *CV B*) is selected, and it is indicated by a blue *Stage LED* (A.3). It simply plays the stage value (whether selected or randomly shifted) for the whole stage duration. It is the normal behavior you would expect from a traditional sequencer.

The second CV *Stage Color*, called *Slide*, is accessed by pushing the *Stage Encoder* (A.1) once, and it is indicated by a green *Stage LED*. In this mode, instead of playing the defined stage value, USTA will automatically generate a linear ramp from the previous value to the new one, in a sort of "glide" effect, interpolated for the whole length of the stage.

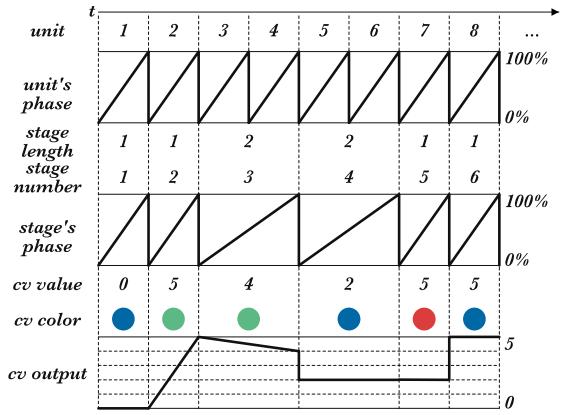


Figure 56: Unit phase, *Stage* phase, and CV output.

The third CV stage color, called *Skip*, is accessed by pushing the *Stage Encoder* a second time, and it is indicated by a red stage LED. In this mode the stage skips the voltage generation, retaining its length (see below): USTA will simply play the last value generated by the previous stage.

Push the *Stage Encoder* again to return to the blue color layer.

Practice the CV Layers and CV Colors with these Techniques:
CV Colors
CV Variation

3.4.4.1 USTA Slide vs FALISTRI Slew

The slide color may be similar to a slew limiter circuit (see FALISTRI §3.3), but it is very different in design and concept. A slew limiter integrates two values and generates an ascending or descending ramp with a fixed duration (lag). Such lag time is arbitrarily set and often allows the target CV to be noticeably performed by the machine. USTA's slide, on the other hand, automatically calculates the lag time, which corresponds to the whole stage duration. This means that the stage value set in the blue layer is technically never played, becoming the

target point of an ascending or descending CV from the previous stage value. Please note that if the gate value is shorter than the stage length, the slide will not be heard in full. This allows for more subtle nuances while composing and can create some interesting results while blended with peculiar Gate settings: a fully open gate will add more dynamic to the composition, while a ratcheting effect will create an ascending sequence of shorter notes, similar to a complex embellishment.

3.4.5 Gate Layers

The other two outputs per track (E.3, E.4, E.7, E.8, E.11, E.12, E.15, E.16) are responsible for the gates. Just like in the CV channels, each stage can have three different *Layers* and three different *Gate Colors*. The first layer determines the number of gates that USTA will play within each stage, while the second and third layers add various degrees of variation to the red layer's value, similar to the CV.

Sets the gate value.	Sets the chance to variate the defined stage's gate number.	Sets the bipolar range for the variation of the gate number.

Table 7: Gate Layers comparison.

3.4.5.1 Red Gate Layer: Value

Figure 57: Gate A's *Value* layer is selected.

The first layer is active by default. Once a gate channel is selected (B.9 or B.10), it is marked by a red RGB LED. It sets the "gate value", which can be either the gate duration (if the stage color is blue) or the number of individual gates (ratcheting) that are generated within the stage length (if the stage color is green). See below §3.4.6 for the *Gate Stage Colors*.

In order to edit the gate number, turn the desired *Stage Encoder* (A.1) clockwise: it will select a value from 1 to 16, which is displayed by the Stage Arc surrounding the encoder. It is possible to set the gate number to 0: in this case, no gate will be generated, and the output will remain low for the whole stage duration.

3.4.5.2 Green Gate Layer: Variation Index

Figure 58: Gate A's *Variation Index* layer is selected.

The second layer is accessed by pushing the channel button (B.9 or B.10) a second time and it is represented by a green LED: it controls the probability that USTA will randomly change the gate number, according to the range set in the blue (third) layer.

Gate Variation Index works similarly to the CV variation index layer (see above): the *Stage Encoder* (A.1), when rotated clockwise, will increase the chance that USTA will change the value selected in the red layer. By default, no LED is lit in the *Stage Arc* (A.2), therefore there are no chances that USTA will change the gate value. When half of the LEDs are lit, the chances of a variation are 50-50, and when all the LEDs are lit it is 100% sure that USTA will not play the selected gate value: it is still possible, however, that the previously defined gate value will be played again as a result of the coin toss, being it still within the *Variation Range* (see below).

3.4.5.3 Blue Gate Layer – Variation Range

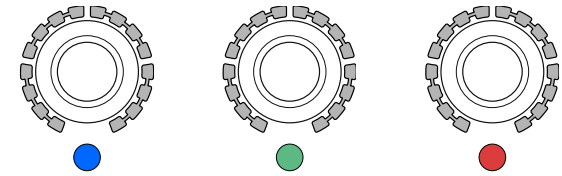
Figure 59: Gate A's *Variation Range* Layer is selected.

The third layer is accessed by pushing the channel button (B.9 or B.10) a third time and it is represented by a blue LED: it determines the range of values that USTA will consider when changing the default value after the "coin toss" set by the green layer.

The *Gate Variation Range* layer selects the possible values that USTA can pick to vary the division of the stage length when the second layer probability is higher than zero. 16 numbers are possible, displayed by the *Stage Arc* (A.2) around the encoder. The default value is 0 (no LEDs are lit).

For instance, if the gate number in the red layer is 2, and a variation range of 8 is selected, USTA will pick a pseudo-random number between −6 (2-8) and 10 (2+8). Every negative result corresponds to a gate-off (i.e. a pause).

3.4.6 Gate Stage Colors



The Stage Arc shows how long the gate stays high	The Stage Arc shows how many gate events are generated	No gate events are generated.

Table 8: Gate Colors comparison.

The first *Stage Color* is called *Gate Length*. *Gate Length* is available by default once a channel (*GATE A* or *GATE B*) is selected, indicated by a blue *Stage LED* (A.3). In this mode, the encoder sets the gate length, which is the portion of the overall stage in which the gate stays high. 17 lengths are available, indicated by the 16 LEDs of the Stage Arc. If no yellow LEDs are lit, the gate will remain low for the whole stage duration, which is the equivalent of a musical pause. By rotating the encoder clockwise, the gate length will be progressively increased, until all the 16 yellow LEDs are lit: in this position, the gate will remain high for the whole stage duration and it will be tied to the next stage's gate. With the blue stage color selected, the Variation parameters set by the blue and green layers will modify the gate duration.

The second *Stage Color*, called *Gate Number*, is accessed by pushing the *Stage Encoder* (A.1) once, and it is displayed by a green *Stage LED*. In this mode, the encoder sets the number of gate events that will be generated within the overall stage length, which is basically a ratcheting effect. Again, 17 options are available, displayed by the 16 LEDs of the *Stage Arc*. If no yellow LED is lit, the gate will remain low for the whole stage duration, which is the equivalent of a musical pause. By rotating the *Stage Encoder* clockwise, the gate value will be progressively increased, until all the 16 yellow LEDs are lit. 1 LED means that 1 gate will be generated, and so on until 16 fast gates are outputted.

This division is performed by USTA based on the stage duration set by the length parameter, therefore, if the gate number is 2, the individual gates will have different spacing and duration according to the stage length.

With the green stage color selected, the Variation parameters set by the blue and green layers will modify the gate numbers, thus creating a different number of gates (ratchet pattern).

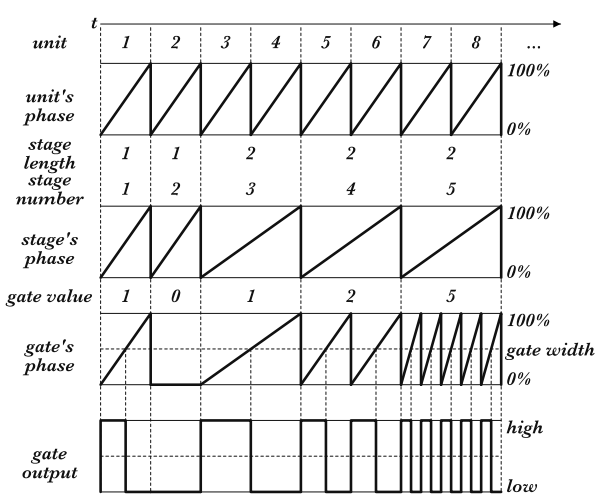


Figure 60: Gates in *Repeat* (green *Color*).

Practice the Gate Layers and Gate Colors with these Techniques:
Gate Colors
Gate Variation

By default, with the green color the gate length is the 50% of the time between two consecutive rising edges; it is possible to change this setting in the *Track Menu* (see above, §3.2), under the `Gate Width %` option.

The third *Color*, called *Skip*, is accessed by pushing the stage encoder a second time, and it is displayed by a red *Stage LED*. In this mode USTA will not generate any gate signal for the whole duration of the stage: it is the equivalent of a musical pause.

An identical effect can be obtained by setting the *Gate Value* to 0 (= no led on the LED Arc) either in the blue or green color; the *Skip* color, however, is useful in case you want to momentarily de-activate a stage while performing, while still retaining its specific *Gate Value*.

4 QUICK EDITING

Here are the functions that USTA provides to facilitate the compositional process. They are achieved through the three grey buttons to the left of the display, called *Mod Buttons*.

4.1 SET ALL AND SHIFT ALL

When in edit mode, the first two buttons allow you to edit multiple stages at the same time. Push them before rotating the stage encoders and hold throughout the operation.

The topmost button, called *Set All* (B.1), allows you to set all stages in a pattern to the value of stage being edited, regardless of their original values. It can be held while editing values (rotating the *Stage Encoder* – A.1) or colors (pushing the *Stage Encoder*), for CVs, Gates, and *Length*. It is also possible to hold the *Set All* button while setting *Track Menu* options through the *Navigation Encoder* (B.14): in this way, the setting will be applied to all the four tracks at the same time.

The second button, called *Shift All* (B.2), works similarly, but shifts all the stages by the same amount as the one being edited, thus transposing their values. Like *Set All*, this button works both for values (rotating the *Stage Encoder*) and colors (pushing the *Stage Encoder*). While shifting colors, all the following stages will change from one color to the next one relative to their previous setting.

It is possible to change the *Shift All* behavior in the *Project Menu* (see above, §3.1) through the `All Edits` option: select `From` to edit all the stages after the one which is being edited (it is particularly useful for quickly writing melodic lines); select `All` to edit all the stages of a pattern, including those before the one being edited.

4.2 Coarse and Fine

In the *Value* CV layer (§3.4.3.1 above) it is possible to edit the stage values by increments of 1 octave or 1V (if the selected channel is working in *Pitch* or *Raw* voltages respectively): to do so, push and hold the *Coarse* button (B.3) while rotating the *Stage Encoder* (A.1) (see Figure 61).

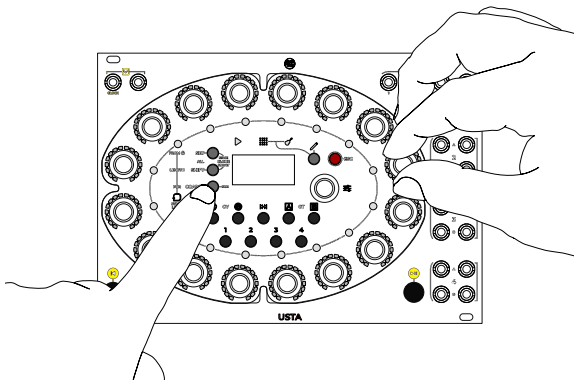


Figure 61: *Coarse* editing.

It is also possible to fine-tune the *Pitch* and *Raw* voltages. Hold pressed *Esc* (B.4) then the *Coarse* button to access the *Fine* mode: now the *Stage Encoders* will change the stage value by cents of semitone in pitch mode, and by milli-Volts in raw mode.

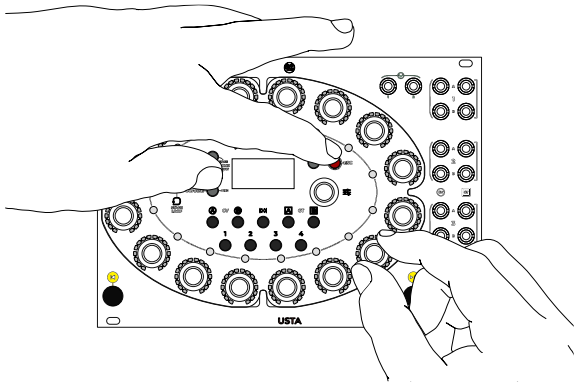


Figure 62: *Fine* editing.

4.3 Combining the Buttons

It is possible to use the *Coarse/Fine* button (B.3) together with *Set All* (B.1) and *Shift All* (B.2) in order to perform tasks such as transposing the whole pattern by an octave or fine-tuning all the stages at once.

Practice the Mod Buttons with these Techniques:
Set All and Shift All
Coarse and Fine Tuning

5 PERFORMING

Once everything is set up according to your musical needs, you can start to perform with USTA.

5.1 Play/Pause, Reset and Master Track Settings

In order to perform with USTA you need to start, pause and reset your sequence. Such controls are deeply connected and rely on the *Master* and *Slave* relationship between *Tracks*.

5.1.1 Master Track

The *Master Track* is the track which other tracks refer to for play, pause and reset purposes. The *Master/Slave* relationship is limited to these three operations: every other parameter such as tempo and time ratio will remain independent per each track. There can be only one *Master Track* for every project loaded into USTA.

To set the *Master Track*, enter the *Project Menu* by holding down the Navigation Encoder for three seconds (B.14), scroll to the `Master Track` option and then select one of the four tracks. By default, a project's *Master Track* is track 1.

To subordinate a track to the *Master Track*, enter the *Track Menu* by pushing the *Navigation Encoder*, scroll to the `Reset On` option and set it on `Master`. By default, all the tracks are *Slave* to the *Master Track*. The *Master* and *Slave* tracks together are called *Master Track Group*.

Once the *Master Track Group* is defined, the *Play/Pause* and *Reset* commands (see below, §§5.1.2-5.1.3) will affect all the tracks, no matter which one is currently selected.

It is possible to detach a track from the *Master Track*: with this option, the *Play/Pause* and *Reset* commands performed within the *Master Track Group* will not affect this track, and vice versa. In case more than one track is detached from the master track, the *Play/Pause* and *Reset* commands will affect them all, whenever they are performed on one of these tracks.

To detach one or more tracks from the *Master Track*, select either the `Local` or `Instant` reset option in each track's `Reset On` menu item.

5.1.2 Play/Pause

In order to start or arrest your sequence, push the *Play/Pause* button (B.13). By default, this button works as a global control: if at least one track is playing, it pauses it; if all the tracks are playing, it pauses all of them; if all the tracks are paused, it plays all of them.

It is possible to play or pause only the selected track by holding *Esc* (B.4) before pushing the *Play/Pause* button.

Additional operations can be performed by combining the *Play/Pause* button with the *Set All* (B.1) and *Shift All* (B.2) buttons.

If the selected track belongs to the *Master Track Group*, pushing *Play/Pause* while holding *Shift All* will play or pause all the tracks of the group. In the case at least one

track is paused, this combo will pause all the other ones within the group.

Hold *Set All* to set all the tracks of the *Master Track Group* like the one currently selected: if it is paused, all the related tracks will pause; if it is playing, all the related tracks will play. Each modification will happen after the current *Master Track* stage has ended.

If the selected track does *not* belong to the *Master Track Group*, pushing *Play/Pause* while holding *Shift All* will play or pause all the tracks which are unlocked from the Master Track. In case at least one track is paused, this combo will pause all the other ones outside the *Master Track Group*. Hold *Set All* to set all the non-*Master* tracks like the one currently selected, as described above for the *Master Track* group. Each modification will happen according to the reset setting of the track currently selected: if `Local`, it will happen after the current stage of the selected track has ended; if `Instant`, it will happen instantaneously.

The *Play/Pause LED* (C.1) will display the current state of the selected track: red if paused, green if playing.

5.1.3 Reset

The *Reset* button (B.12) restores the playhead at the beginning of a pattern, i.e. at stage 1. It is possible to set whether the playhead returns to the first stage of the current pattern (*Reset Stage*) or to the first stage of the first pattern (*Reset Stage&Pattern*). To change this parameter, enter the *Track Menu* (see above, §3.2), scroll until the `Reset What` option, and select either `Stage` or `Stage&Pa`. It is also possible to select the option `Nothing`, thus disabling the *Reset* button.

Each track can respond to the *Reset* command in four different ways. The first two depend on the track's relationship with the *Master* track:

· If the selected track is the *Master* track, the *Reset* button will reset it once the current stage has ended;

· if the selected track is a *Slave* track, the *Reset* button will reset it once the current stage of the *Master* track has ended: both these options are achieved by entering the *Track Menu*, scrolling to the `Reset On` option and setting it on `Master`. This option is useful if you need to force several tracks to reset at the same time (for instance, if you are playing with different tempos or ratios).

If you want to reset a track independently, you have two more settings of the `Reset On` option, called `Local` and `Instant`:

· `Local` resets the selected track once its current stage has ended, without resetting any of the *Master/Slave* tracks;

· `Instant` resets the current track instantly, without waiting for it to end the current stage.

The *Reset* function can be executed through external trigs, see below §7.2.

When *Pattern Shift* is enabled for a track (see below, §), the Reset button will set the sequence back to the first pattern plus the pattern shift number determined by the *CV offset*.

5.2 Stage Loop

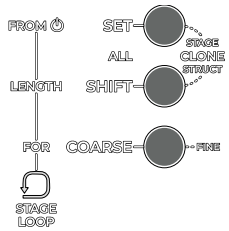


Figure 63: The *Stage Loop* section and controls (on the left).

In addition to the options above, USTA gives the chance to loop a specific portion of your pattern structure independently for each track through the *Stage Loop* option. Such a loop is nested within the main play mode and can work across different patterns. The looping function works only in *Pattern Mode*, and it interacts both with *Full Pattern Recall* and *Pattern Mix* (see below, §5.4). Once Song Mode is activated, the *Stage Loop* function will be automatically disabled.

Three variables are used to control the loop behavior: `FROM`, or the stage/pattern from where the loop starts; `LEN` (*Length*), how many stages are involved in this loop, and `FOR`, how many times it loops. These variables are shown in the bottom line of the *Dashboard* (F.17, F.18, F.19).

To define the *Stage Loop* `FROM`, hold the *Set All* button (B.1) and rotate the navigation encoder. The first number is the stage number, and the second one the patter number.

To define the Stage Loop `LENGTH`, hold the Shift All button and rotate the navigation encoder – length is expressed stages, so the resulting length depends on each stage length. Values go from 1 to 16.

To define the Stage Loop `FOR`, hold the Coarse button and rotate the navigation encoder – the `FOR` parameter express how many times these stages are looped until exiting this loop. Values go from 0 to 16 where 0 is no loop at all.

The only anchor point of the Stage Loop is the `FROM` parameter: it does not have a specific endpoint, but it is instead defined by a length which is expressed in number of stages.

To enable or disable the *Stage Loop*, double click the *Set All* button (B.1). When *Stage Loop* is active, the Stage Loop LED (C.6) will light up: if the playhead has not yet reached the *Stage Loop Section*, it will be red; if the playhead is already within the loop section, it will be green.

Stage Loop is disabled for the selected track	Stage Loop is enabled for the selected track, but the playhead is not yet in the loop	Stage Loop is enabled for the selected track, and it is in the loop

Table 9: Stage Loop LEDs comparison.

When *Stage Loop* is active, USTA will start the loop section once the playhead reaches the *From* point, it will play the stages within the *Stage Loop* for the defined number of times, then it will continue with the original sequence.

A small example with the following settings: `FROM02/03`, `LENGTH 04`, `FOR 02`. This means that on Pattern 3 it will play the following stages: 1 2 3 4 5 [2 3 4 5 2 3 4 5] 6 7… The stages between square brackets are the *Stage Loop*.

5.2.1 Infinite Stage Loop

If you set the `FOR` value to 0, the *Stage Loop* will loop endlessly once engaged: in this way, its duration is completely up to you. To make the playhead "exit" the *Infinite Stage Loop*, double click the *Set All* button (B.1) to disable the *Stage Loop*.

Practice the Stage Loop:
Stage Loop

5.3 Song Mode

The *Song Mode* allows you to arrange your track's patterns in a custom sequence. It is an alternative mode of playing beside the default *Pattern Mode*: this means that *Song* and *Pattern Mode* cannot be active at the same time, but an option in the Track Menu lets you choose which one USTA will follow during the playback.

The *Song Mode* is a series of patterns that are located into *Slots*. A song can host up to 4 *Pages* of 16 *Slots* each for a total of 64 *Slots*. One song per track is available.

Once at least two patterns are composed in one track, the song mode allows you to arrange them in the desired order by fitting each of them into one or more slots. Each slot can be repeated up to 16 times before moving to the following one.

Bear in mind that the song is not a structural part of the track but an alternative arrangement of items, therefore, it cannot be cloned.

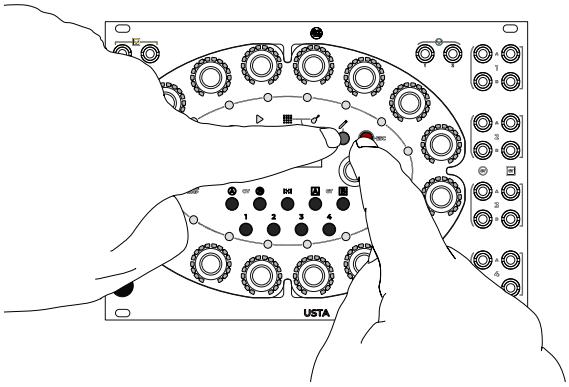


Figure 64: *Edit Song* combination.

You can create a *Song* while USTA is still playing in *Pattern Mode*. To do so, enter *Edit* mode (if you are not there yet) by pushing the *Pencil* button (B.5) and then enter *Edit Song Mode* by holding *Esc* (B.4) and pushing the *Pencil Button* again.

Now the guitar-shaped *Song LED* (C.3) will light up red, and the *Patten LED* (C.2) will be green: this means that you are editing the song, while still being in *Pattern mode*.

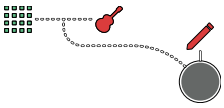


Figure 65: Editing a song wile in *Pattern Mode*.

In this configuration, each of the 16 *Stage Encoders* (A.1) edits a slot of the selected page by performing two operations: selecting the pattern to be played in the desired slot and selecting the number of times that such pattern must be played (0 to 16). To change *Page* simply rotate the navI

Once entering the Edit Song mode, all the five Parameter LEDs will turn red: this means that now USTA is editing the pattern number per slot.

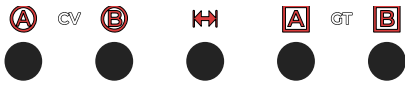


Figure 66: The LEDS when editing the pattern number per slot.

Rotate each encoder to select the pattern to be played, numbered 1 to 32: the *Stage Arc* (A.2) will display the selected pattern number.

To edit the number of times that a pattern should be repeated per each slot, push any of the Layer buttons to access the repetition parameter: the Layer RGB LEDs will all light up green.

Figure 67: The LEDs when editing the repetition number per slot.

In this mode, each encoder sets the number of repetitions, from 0 to 16, and the *Stage Arc* will light up accordingly.

To switch from *Pattern* to *Song* mode, enter the *Track Menu*, scroll to the `Play Mode` option and change the setting from `Pattern` to `Song`. Exit the menu, and you will notice a different layout of the LEDs:

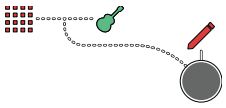


Figure 68: Editing a song while in *Song Mode*.

This will indicate that the selected track is currently in *Edit Mode* (red pencil LED – B.5), it is in *Song Mode* (green guitar LED – C.3), and you are seeing/editing the patterns (red pattern LED – C.2).

Once your song structure is set, push the *Play/Pause* button (B.13) to start the playback: while playing, the playhead will display and behave differently depending on the mode selected.

If the track is in *Edit Song Mode*, the stage LED:

- will be blue when that slot has a repetition bigger than zero,
- will be red only on the current playing slot
- will be cyan on the stages that are playing on that pattern. In the example below slots 1, 2, 3, 4, 7, 8, (maybe 9) and 16 has a length, the song is playing the 3rd slot, and in that pattern, the 9th stage is playing;

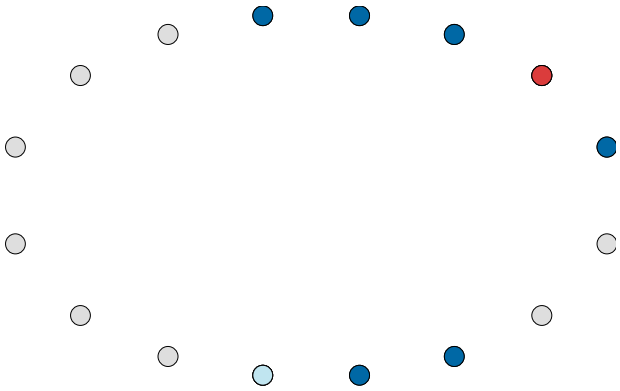


Figure 69: Stage LEDs in *Edit Song*. Note the playhead on Stage 9.

If the track is in *Edit Pattern Mode*, the playhead will follow the rules of this mode as described above (see Editing, Playing and Looping Patterns): it will be visible only when USTA plays the pattern that is currently being edited;

If USTA is in *Performance Pattern Mode*, the playhead will move through all the steps of all the patterns following the structure determined by the Song.

When the *Song Mode* is enabled, the *Pattern Loop* indicators on the dashboard will be replaced by `Rep` followed by two numbers: these stand for 'repetition' and display the current repetition of the pattern/the total number of repetitions set for that slot.

5.3.1 Pattern to song while playing (and vice versa)

It is possible to play in pattern or song mode, but it is also possible to change mode while playing without stopping the track or losing its sync: in the *Track Menu*, change the *Play Mode* option, and at the end of the pattern the selected mode will come into play.

Get an overview of the *Song Mode* with this Technique:
Song Mode

5.4 Live Performance Tools: Pattern Recall

When USTA is in Performance mode, either in *Pattern* or *Song* mode, every editing option is disabled: the 16 *Stage Encoders* (A.1) are now associated with the first 16 patterns, and while holding the *Shift All* button (B.2) they are associated with the last 16 patterns (or from the 17th to 32nd).

They work only as buttons, by pushing which it is possible to recall any of the patterns on the fly, thus temporarily bypassing the sequence of patterns you are expecting from the Pattern Loop rules or Song Mode set above. Once the selected pattern has been recalled and played, USTA will get back at the original sequence, with different outcomes according to the play mode in use and the pattern loop selected.

USTA will memorize only one value: it means that if you push two encoders during this mode, only the latter will be effective.

Furthermore, there are two options: *Full Pattern Recall*, which is the default one, or *Pattern Mix*, available by holding down *Coarse* (B.3) before pushing the pattern encoder.

These recall options affect the selected track only.

5.4.1 Full pattern recall

With the *Full Pattern Recall*, USTA will wait until the active pattern has reached its end, then it will play the one selected by pushing its *Stage Encoder* (A.1).

If the selected patter is *within* the *Pattern Loop*, USTA will continue to play the patterns after the selected one. (For example, if the *Pattern Loop* lasts from pattern 1 to 10, and you recall pattern 6 while being in pattern 2, the sequence would be: 2, 6, 7…).

If the selected pattern is *outside* the *Pattern Loop*, USTA will play it once, then continue after the previous pattern. (In the same example as above, if you recall pattern 15, the sequence would be: 2, 15, 3, 4…)

5.4.2 Pattern Mix

The *Pattern Mix* function is similar to the Full Pattern Recall, the main difference being when the pattern change happens. It is achieved by holding *Coarse* (B.3) before selecting the pattern through the *Stage Encoders* (A.1– Remember to hold *Shift All* to select patterns no. 17-32). Once the new pattern is selected by pushing the encoder, the playhead will wait for the playing stage to end and play the consecutive stage on the selected pattern, instead of the current one. For example, if you are playing stage 7 of the second pattern and select the fifth pattern, USTA will wait until the end of stage 7, then move to stage 8 of pattern 5, mixing them.

If the selected pattern does not have enough stages to maintain the sequence length of the current pattern, or if it is selected during the last stage of the current pattern, USTA will play the selected pattern starting with its first available stage.

To sum up, *Full Pattern Recall* juxtaposes patterns, while *Mix* combines them.

Practice difference between these two functions with this Technique:
Pattern Recall and Pattern Mix

5.5 Mute

One of the most popular features of live performances is the *Mute* function, i.e., being able to immediately remove or add certain parts of the composition. USTA features two levels of muting: *Mute Track* and *Mute Channel*.

5.5.1 Mute Track

This function temporarily disables all the outputs of the muted track.

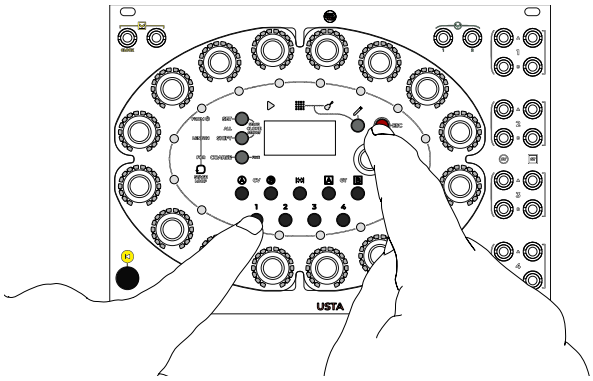


Figure 70: *Mute Track* button combination.

Muting a track results in it having no Gate A and B output and no *CV A* and B output (all 4 outputs of the muted track provide a constant 0V). To achieve this function, hold the *Esc* button (B.4) and push the button of the track you want to mute (B.11 see Figure 70). It is possible to mute any desired track, even it is not the currently selected one.

A muted track shows the `TRACK MUTED` text on the current stage section of the *Dashboard* when selected.

5.5.2 Mute Channel

On a more specific level, the *Mute Channel* function allows to mute solely the output of one or more channels per each track (*CV A*, *CV B*, *Gate A*, *Gate B*). This option may be useful in case you are using different channels of the same track to control different sound sources, such as *Gate A* for the kick drum and *Gate B* for the snare.

To mute a channel, hold the *Esc* button (B.4) and push any *Channel Button* (B.6, B.7, B.9, B.10).

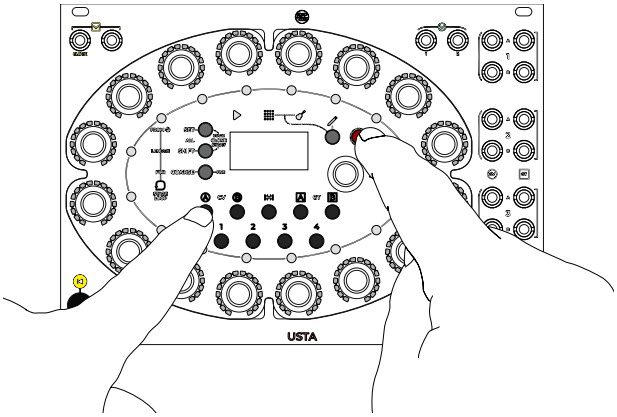


Figure 71: *Mute Channel* button combination.

5.6 Hold

It is also possible to "freeze" the current channel value in a specific track through the *Hold* function: by holding *Set All* (B.1) and pushing one of the *Channel Buttons* (B.6, B.7, B.9, B.10), USTA will hold the values as they are playing in that specific moment until the buttons are released, see Figure 72.

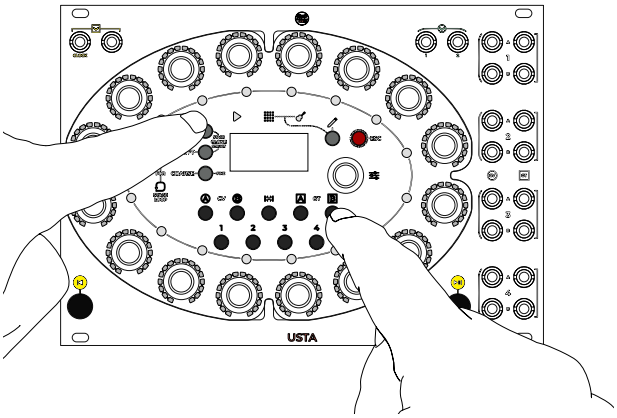


Figure 72: *Hold* button combination.

If the selected channel is *CV A* or *CV B*, USTA will hold the given value (i.e. it will not calculate other stages' values); if the selected channel is *Gate A* or *Gate B*, USTA will perform a sort of legato (if the gate is high when the channel button is pushed) or mute (if the gate is low).

After releasing the buttons, the sequence will return to playing from the position reached by the playhead in the meantime.

6 ADVANCED EDITING

6.1 Composition Mode

If you prefer to compose when USTA is not playing, e.g. if you have to handle longer polyphonic sequences, crossing several patterns on multiple tracks, USTA provides a system for asynchronous voltage monitoring called *Composition Mode*.

When USTA is in *Edit Pattern Mode* and paused (red *Play/Pause LED* – C.1), select the CV layer you want to edit and double-click the desired track button: this will set *Gate A* and *Gate B* high, and the cyan playhead will light up on the last played stage. In case the last played stage is in a different pattern than the selected one, the first stage of the selected pattern will light up cyan and open its gate.

Rotate the *Navigation Encoder* (B.14) back and forth to move the playhead across the different stages: this will allow you to manually "scan" through the sequence in order to check and edit the current stage settings. This parameter will work in both directions just like an analog reel-to-reel recorder, and it will work through different patterns: once the 16th stage is reached, further clockwise rotation of the *Navigation Encoder* will move the playhead to the first stage of the following pattern, while rotating counterclockwise past the first stage of a pattern different than the first move the playhead to last stage of previous pattern.

The display will automatically update the stage data (*CV A*, *CV B*, *Gate A*, *Gate B*, and *Length*) according to the selected stage.

Once you have finished your editing operations, double-click again the selected track: USTA will get back to the original sequence and it will play it starting from the last played stage (i.e., before entering the Composition Mode). In case you want to hear your sequence from the beginning, simply push the *Reset* button (B.12).

If you push the *Play/Pause* button (B.13) while still being in *Composition Mode*, the sequence will play after the one currently selected. This feature can be very useful in case you want to hear portions of your sequence starting in other stages than the first one.

Composition Mode is not available when the *Track Menu* is open, and it ignores Green or Red CV/Gate stage colors, being Gate on that track always high and CV always reproducing exactly the selected stage value.

Practice this mode with the following Technique:
Composition Mode

6.2 Use an External CV/Gate Keyboard

In order to compose your sequence with an external keyboard (or any other 1V/oct controller), you need the current track to be stopped (red *Play/Pause LED* – C.1) and in *Composition Mode*.

Connect the keyboard gate output to the *Auxiliary Gate* input (D.2) and the keyboard CV output to the *CV A* input (D.3). From now on, pushing a key on the keyboard will overwrite the defined note (*CV A*) of the stage highlighted in composition mode, and the keyboard gate out coming from the key release will move the playhead to the following stage.

This method inputs only the pitch values, regardless of the stage length: to set the rhythm, refer to the section about setting the *Length*, see above §3.4.1).

Bear in mind that when composition mode is enabled, the main function of the *Auxiliary Gate* input is bypassed to allow the use of the external keyboard composition mode.

Practice with this Technique:
Keyboard Input

6.3 Store Pattern: Last Played or Last Full

Should you prefer a less deterministic compositional approach, e.g., by progressively adding some degree of variation through the green and blue layers, while your sequence is playing, or adding external modulations via CV, until you find the perfect combination, USTA stores the last played values of a pattern, allowing you to recall and reproduce them.

Every time USTA plays a stage, its values are stored in volatile memory until the playhead crosses it again, overwriting it. This leads to a constantly updating 16-element array, one per stage, per each track, which can be transferred anytime over the desired pattern.

The data stored per last played stage are the values for all the *Channels* (*CV A*, *CV B*, *Gate A*, *Gate B*) as well as the *Length*: no data about *Variation Index*, *Range*, or *Stage Colors* is stored.

There are two store modes, *Last Full Pattern* and *Last Played*. The difference between the two is that former updates the full array at the end of the pattern, while the latter updates the array every time a new stage is played, while removing the oldest array element.

To access the stored values, select the track (B.11) whose stages you want to copy through the layer buttons, make sure you are in *Edit Mode* (*Pencil* button and LED, B.5) select the target pattern by rotating the *Navigation Encoder* (B.14 – you can also leave the current pattern and overwrite its values while playing), then push the *Shift All* button twice (B.2) for the last full pattern, or the *Coarse* button twice (B.3) for the last 16 played stages: the result is now dumped in the desired pattern.

These actions work also when USTA is stopped, in case you want to dump multiple layer values or make more "copies" of a pattern. Every value is linked to its respective stage and will be dumped only on it.

All the values are set to 0 at start up. If you overwrite the current pattern, all the previous data concerning *Variation* (green and blue layers) will not be affected, so the resulting sequence might differ from the one you copied due to the *Variation* still being performed by USTA: if you want to hear the last played sequence, simply set all the *Variation Index* and the *Variation Range* to 0. The same approach is applied to possible CV modulations in use. If you perform the dump halfway through a pattern, which has for example stages 8-16 with length 0, or if the store is performed straight after the start-up, the dumped value of the non-played stages will be 0.

6.4 Rotate Pattern

It may happen that, when improvising with various degrees of unpredictability, you end up with the perfect sequence, only to find out that it is one or two stages offbeat. To overcome this issue, USTA allows you to *Rotate* a pattern and align it to the other tracks, so that the beginning of your newly found sequence is exactly where it is supposed to be.

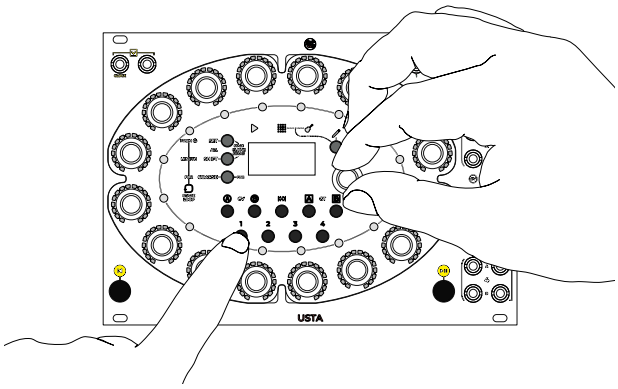


Figure 73: *Rotate Pattern* button combination.

To *Rotate* a pattern, push and hold the *Track* button (B.11) and rotate the *Navigation Encoder* (B.14) either clockwise or counter-clockwise: you will see the 16 stages rotate accordingly, and, if your sequence is running, you will hear it changing immediately (see Figure 73).

This function shifts the *Length*, *Channels* and *Layers* of a pattern at the same time (all the stage values, with their *Variation Index* and *Range*).

6.5 Cloning

6.5.1 Clone a Stage

It is possible to clone all the data of a specific stage onto another of the same pattern. By doing so, all the layers of the target stage (see above the section concerning editing stages, §3.4) will be replaced with the data contained in the source.

To do so, hold the *Esc* button (B.4) and push *Set All* (B.1), then release both buttons.

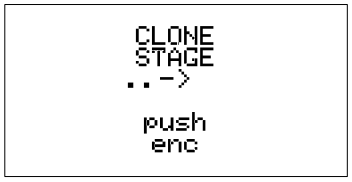


Figure 74: Cloning a stage.

The *Clone Stage* menu will appear in the OLED display (C.5). First, enter the stage to clone: push the corresponding stage encoder (A.1, no. 12 in the example): its number will appear within the display.

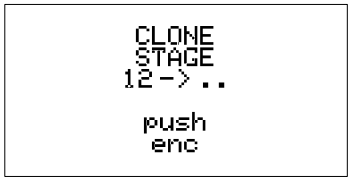


Figure 75: Cloning stage 12.

From now on, push any other *Stage Encoder* to clone the data of the source stage. This last operation can be repeated as many times, to clone a single stage towards multiple destinations.

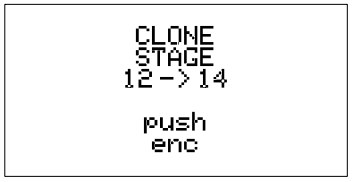


Figure 76: Cloning stage 12 to 14

The last cloned target will be updated on the display, as above (14 in the example).

At any moment push *Esc* to exit the *Clone Stage* function.

6.5.2 Clone a Structure

It is also possible to clone structures (STRUCT) of stages: these are in fact *Layers*, *Patterns*, and *Tracks*. Layers and patterns may be cloned within their track only.

To clone any of these *structures*, hold the *Esc* button (B.4) and push *Shift All* (B.2), then release them.

```
        CLONE
LAYER PATTERN TRACK
..->   ..->   ..->

layer+ push   push
 push  enc   track
 enc
```

Figure 77: Clone Structure selection.

The *Clone Struct* menu will appear in the display, proposing all structures that can be cloned. At any moment simply push *Esc* to exit the *Clone Struct* function.

6.5.2.1 Clone a Layer

```
CLONE
GATE A
12->..

push
enc
```

Figure 78: Cloning Stage 12's Gate A value.

To clone a *Layer* (*CV A*, *CV B*, *Length*, *Gate A*, *Gate B*) of a pattern, push and hold the button for the layer you want to clone (B.6-B.10), then push the *Stage Encoder* (A.1) to define the source pattern (encoder 1 for pattern 1, encoder 2 for pattern 2 and so on). To access patterns from 17 to 32, push and hold also the *Shift All* button (B.2).

From now on, push any other *Stage Encoder* to clone the selected layer of the source pattern to the desired target (the example shows cloning of *Gate A* data of pattern 12 of the selected track). Press other pattern encoders to repeat the cloning to multiple targets without restarting the procedure.

```
CLONE
GATE A
12->14

push
enc
```

Figure 79: Cloning Stage 12's Gate A value to Stage 14.

The last cloned target will be updated on the display as above (14 in the example).

6.5.2.2 Layer Cross-Cloning

It is also possible to clone a layer to a different CV or Gate channel than the original one, for example, cloning the *CV A* Value of Pattern 1 to *CV B* of pattern 3, or the Gate B Variation Index of Pattern 15 to Gate A of Pattern 22.

To do so, follow the same procedure as for cloning a layer, but hold the target CV or Gate button (B.6, B.7, B.9, B.10) before pushing the target Pattern encoder.

6.5.2.3 Clone a Pattern

To clone a *Pattern*, when the *Clone Struct* menu is displayed, push the *Stage Encoder* (A.1) for the source pattern (encoder 1 for pattern 1, encoder 2 for pattern 2 and so on). For patterns from 17 to 32, first hold down the *Shift All* button (B.2).

```
CLONE
PATTERN
12->..

push
enc
```

Figure 80: Cloning Pattern 12.

From now on, push any other *Encoder* to clone the data of the source pattern to the desired target (the example shows cloning pattern 12 of the selected track). Push additional pattern encoders to clone to multiple targets without restarting the procedure.

```
CLONE
PATTERN
12->14

push
enc
```

Figure 81: Cloning Pattern 12 to Pattern 14.

The last cloned target will be updated on the display as above (14 in the example).

6.5.2.4 Clone a Track

To clone a *Track*, when the *Clone Struct* menu is displayed, select the source track by pushing its corresponding *Track Button* (B.11).

```
CLONE
TRACK
1 ->..

push
track
```

Figure 82: Cloning Track 1.

From now on, push any track to clone the data of the source track to any target track you want (the example shows cloning of track 1).

```
CLONE
TRACK
1 -> 2

push
track
```

Figure 83: Cloning Track 1 to Track 2.

Press additional track buttons clone to multiple targets without restarting the procedure.

The last cloned target will be updated on the display as above (2 in the example).

Keep in mind that all clone functions are destructive editing and are similar to edit the stage with their encoders.

6.6 Quick Track Initialization

Composition is a process that involves a certain amount of trial-and-error. In case you are not satisfied with what you came up with, you can always erase the content of a track, without having to manually do it, or worse, to start a new project from scratch.

To initialize a track, select it (B.11), make sure to be in *Edit Pattern* mode (see above, §3.3), and then push and hold the *Pencil* button for three seconds (B.5).

This procedure will restore the selected track to its original state, without affecting the other three.

6.7 Quick Song Initialization

On the same line, it is also possible to initialize a *Song.*, in case you want to quickly arrange your patterns in a new order.

To do so, select a track, then enter *Edit Song* mode (see §5.3); from here, push and hold the *Pencil* button for three seconds (B.5), and you will restore all the song *Slot* values to 0.

7 EXTERNAL CONTROLS

Four jack sockets on the top of USTA's front panel are dedicated to external modulation: two are designed to receive external trig or gates, and two for external CVs.

The external gate sockets are used for an external *Clock Input* (D.1) and *Play/Pause* or *reset* (the *Auxiliary Gate*, D.2); the CV inputs (D.3 and D.4) can be individually routed to four different modulations per track, called *Varishift*, *Stage Shift*, *Gate Shift*, *Root Shift*, and *Pitch Shift*. These four modulations are called *Shift* because they all increase (or decrease) a stage's set value according to the incoming voltage. Please note that all these modulations are applied by USTA after a stage has ended, except *Pitch Shift*, which is the only one that can happen in real time, i.e., during a stage.

7.1 Clock Input

To use an external clock source, connect it to the *Clock Input* (D.1) and select `External` from the `ClockSource` option in the *Track Menu* (see above, §3.2.1). USTA will now operate just like with its internal clock, with some minor considerations.

When an external clock is in use, the options of *Swing* (see below, §8.4) and *Ratio* (see above, §3.2.1) are always available: however, since they are calculated by USTA for each new stage over a median value of the previous impulses, the quality of the effect depends on a steady external clock.

In case Swing and ratios different than 1:1 are not needed, it is possible to use random or even manual trigs to animate the sequence.

7.2 Auxiliary Trig/Gate Input

USTA offers an additional input for external gates (D.2), whose effects can be chosen according to internal routing parameters.

By default, it resets the sequence, but it can also be used to receive *Run* signals to sync USTA with other hardware. This setting is global and can be accessed through the *Project Menu* (§3.1), under the item `Aux Target`.

7.2.1 Reset

When the `Aux Target` is set to `Reset`, the gate patched to the AUX input performs the same function as the manual *Reset* button (B.12), affecting the tracks according to the `Reset What` (`Stage` or `Pattern`) and `Reset On` settings (`Master`, `Local` or `Instant` – see above, §5.1). A gate high signal equals to pushing the *Reset* button, and a gate low signal equals to releasing it.

7.2.2 Run

Any of the four `Run` option of the `Aux Target` menu is designed to control USTA with other devices while using an external clock.

The gate signal patched to the *Auxiliary* input (D.2) now has three functions: play, stop and reset. Since different devices have different sync behaviors, USTA allows for three different options, as described in the table below:

	Gate High	Gate Low
`Run 1`	Reset, then stop	Play
`Run 2`	Play	Reset, then stop
`Run 3`	Stop, then reset	Play
`Run 4`	Play	Stop, then reset

Table 10: Run modes.

7.3 External CV

USTA has two inputs that welcome CV for modulation purposes. The destination of the modulation is routed

through specific menu settings. One input can be assigned to multiple modulation options.

To enable the external CV modulation for a given track, enter the *Track Menu*, and scroll to the *Shift* section. The options available are `PitchShiftA`, `PitchShiftB`, `Root Shift`, `Gate ShiftA`, `Gate ShiftB`, `Stage Shift`, `Vari Shift`, `Phase Shift`, `Pattern Shift`. Choose the selected destination, push the encoder and select the source among `None`, `Ext` *CV A* and `Ext CV B`.

7.3.1 Pitch Shift

Pitch Shift can be performed witch CV layers in *Pitch* mode.

In this mode, the input is continuously sampled and rounded to the smallest interval available in the quantization scale selected (by default it is a semitone, but it can be changed, see below §9.1). Each time the quantized value changes, the modulated track will shift its pitch accordingly, even while the stage is playing.

If the *Variation* layers (*Index* and *Rang*) are used to modify a stage value, this will be taken into account by USTA at the beginning of the stage, before the *Pitch Shift*, and such variation will not be performed again during the stage, even when the external *Pitch Shift* changes value.

The resulting value is then quantized in reference to the *Scale*, *Root* and *Quantization Direction* in use (§9.1).

Example: in C major, a stage with *Variation Range* value 2 (so ±4 semitones) has a normal note *Value* of D. The *Variation Range* coin toss specifies +2 semitones, resulting in a note value of E. If the pitch shifting is 0 at the beginning of the stage, USTA will play an E. If the CV modulates mid-stage from 0V to 0.5V (+6 semitones), the resulting Pitch Shift would change the E +6 semitones to A#, which would then be quantized, according to the *quantization direction* preferences, to B or A.

It is not possible to perform *Pitch Shift* when using *Raw* CV mode: it would make more sense to sum non-quantized voltages outside USTA, using, for instance, a 333 module.

Pitch modulation "à la" mod wheel is suggested to be performed with a 333 as well, to avoid the stepped effect of the dynamic quantization.

7.3.2 Root Shift

This CV destination lets you modify the root of the *scale/mode* selected for the *Dynamic Quantization* (see below §9.1). If, for instance, the selected mode is Locrian, any incoming CV will change the root and create other Locrian modes to which quantize the ongoing melody. It can be very useful to create modal pieces with CV automations: if the same CV is routed both to the *Root Shift* and the *Pitch Shift*, the result will be an exact transposition.

It goes without saying that if the selected scale is chromatic the *Root Shift* will not produce any noticeable effect.

7.3.3 Gate Shift

This parameter uses the incoming external CV to increase or decrease the gate values played by USTA in a given stage. The lower and higher values will still be 0 and 16: this means that if the stage has a defined value of 0 gates (i.e. it counts as a musical pause), a negative offset will not produce any audible effect; the same happens if the stage has a defined value of 16 gates and it receives a positive offset.

It is possible to use this modulation to mute a given stage by providing a negative offset that will set the gate value to 0.

7.3.4 Stage Shift

This modulation shifts the stage values of the four channels of the track while retaining the original stage length defined by the sequence. In other words, while the playhead keeps the rhythmic structure, the melodic content is changed according to other stage values further down the pattern.

When *Stage Shift* is in effect, and any of the four channels are selected, the cyan playhead will display the target stage, i.e. the "shifted" one; when the *Length* parameter is selected, it will display the regular stage sequence.

An example: a pattern is set up with a stage length of 1 unit from stage 1 to 8, and stage length 0 from stage 9 to 16. The normal behavior when playing it in loop is that the stages are played from the 1 to the 8 then back from the 1st and so on. The CVs and gates generated are the values set in their stages. With Stage Shift, the individual length of the stages is unaltered, but their CV and Gate values will be replaced with the one of the target stages, determined by the incoming CV offset. In this case, if a steady offset of 2 is detected, the pattern will have the length of stages 1-8, but the values of stages 3-10.

The stage shift does not cross patterns, so whenever the result of the shifting is bigger than 16, the number is then folded back from stage 1.

The stage shift is calculated on the stage change: this makes possible to obtain shifts, or drunk modes or any kind of strange stage sequence without affecting the rhythmic structure of your composition.

In order to enable the *Stage Shift* mode, select the desired track, enter its menu and navigate until `StageShift`, selecting the CV channel you want to use as a source for that shifting. Either CV 1 or 2 can be used for this purpose.

7.3.5 Vari Shift (Variation Shift)

Another possible destination for external CV is the *Variation Range* for the *CV A* and *CV B*, which is manually set by the blue layer. With this option, it is possible to automatically increase or reduce the spectrum of values in which USTA can pick whenever the CV Variation chance parameter is higher than 0.

As described in *Blue Gate Layer* above, the *Variation Range* is bipolar: a positive CV will increase such range both above and below the stage value, and a negative CV will decrease it.

Please note that if the CV *Variation Chance* (green CV layer) is set to 0, this modulation will not produce any audible effect.

7.3.6 Pattern Shift

It has been said that the default order of patterns in *Pattern* mode can be temporarily altered through Pattern Recall and Pattern Mix or defined in a different structure through Song Mode.

However, it is also possible to automatically recall other patterns though *Pattern Shift*. This function uses an external CV to change the pattern which would naturally follow the current one in the Pattern Loop: it does not recall, therefore, any pattern outside the defined *Pattern Loop*. For the same reason, the *Pattern Shift* is not designed to operate in *Song Mode*.

By default, in *Pattern Mode* USTA moves forward by 1 pattern at a time. *Pattern Shift* adds an offset to the default +1 operation: for example, if a fixed CV offset equivalent to 1 is applied, USTA will skip a pattern out of two (thus moving from 1 to 3 and from 3 to 5); if the pattern loop contains only three pattern, USTA will skip from pattern 1 to 3, then it will loop back to 2, then back again to 1, 3, 2 and so on. If a pattern contains four patterns, and an offset of 3 is applied, usta will seem to keep looping only the pattern on which the Pattern Shift started, because the new offset will force the loop always back to it.

When Pattern Shift is enabled, the Reset button will behave accordingly: instead of resetting the loop to the First Pattern, it will reset to the pattern whose number is defined by first pattern + the pattern shift offset.

7.3.7 Phase Shift

USTA allows you to use an external CV to shift the phase of a track, i.e. its position in relation to the expected timing: a positive voltage will shift a track up to half time unit "behind the beat", thus allowing effects such as rubato or delays.

The great advantage of the *Phase Shift* is that even if it may seem to slow down the track, it does not affect the clock: when the incoming CV goes back to 0, or when the cable is unplugged, the track will immediately get back to the usual position in time.

Interesting effects can be obtained with manual offsets (like 321), envelopes (like FALISTRI) or clocked random voltages (like SAPÈL).

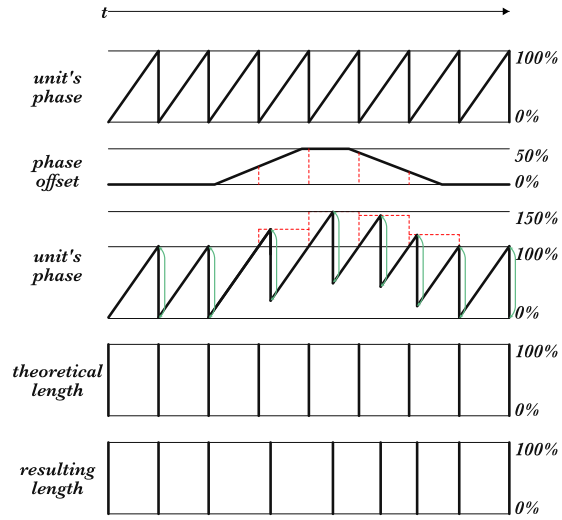


Figure 84: Visual representation of *Phase Shift*.

Practice some of the different CV modulations with these Techniques:
Pitch Shift
Root Shift
Phase Shift

8 ADDITIONAL OPERATIONS

In addition to the previous operations, USTA allows you to perform deeper editing options. According to their position in USTA's architecture, such options are located either in the *Project Menu* (§3.1) or in the *Track Menu* (§3.2).

8.1 Select CV Mode (Raw or Pitch)

In each track, *CV A* and *CV B* can be set to work in *Pitch* or *Raw* mode, independently. These two modes have been created to fulfill any need depending on the addressing of that control voltage: if a control voltage for an oscillator's pitch is needed, or for anything that responds in V/oct, it is suggested to use pitch mode, while Raw mode is suggested for any other application.

By default, *CV A* is set to work in pitch and *CV B* in Raw. To change that, select the track you want to modify, push the *Navigation Encoder* to access the *Track Menu* (B.14), and scroll to reach the *CV A* `Mode` or `CV B Mode` menu item: push again the navigation encoder, scroll to select the desired mode, push again to confirm and push *Esc* (B.4) to exit.

Remind that when *CV A* or *CV B* are set to work in Pitch mode, the dashboard will display the note; when they are set to work in raw mode, the dashboard will display the voltage on a milliVolt scale.

8.2 CV Range

USTA offers the option to set each CV output independently per track to work in unipolar range (from 0V to 10V) or in a bipolar range (±5V). This setting is independent of the CV output being *Pitch* or *Raw*. The reason for this option is that sometimes it might be useful to have negative values for lower notes, such as if you want to tune your oscillators to 440Hz, but some other times it might be useful to have the CV working only into positive values, such as to modify the filter cutoff. By default, every parameter is set to work in a unipolar range.

This setting is 100% software dependent, therefore different settings for each track's channel can be set, stored and recalled for each project.

To set the range of a CV output of a track, push the *Navigation Encoder* to access the *Track Menu* (B.14) and scroll to reach *CV A* `Range` or `CV B Range`, then select the preferred *Range* (`0/10V` or `−/+5V`). Push again to confirm and *Esc* button (B.4) to exit.

If it is necessary to trim a specific CV output, see the section about output trimming.

8.3 Gate Width %

By default, when the *Gate Color* is green (see above, §3.4.6) the gate width (i.e. the time it stays high) is the 50% of the time between two consecutive gate rising edges. It is possible, however, to change this percentage independently for gates A and B for each track. In the Track Menu select *Gate A* `% Width` or *Gate B* `% Width` and choose the desired value: the selectable range spans from 10% to 90%.

8.4 Swing

USTA allows adding a swinging feel to each track independently through the *Swing* option.

Traditionally, swing is a rhythmic style that implies an alteration of the standard eighth note pattern by playing the downbeat note longer than the upbeat one, thus providing a sort of "bouncing" feel. It is commonly played in blues and jazz styles, but it can be a useful tool to add some groove to an electronic composition too.

There is no fixed rule to determine the swing ratio, or the relationship between the first and the second element of a couple of eighth notes played with a swing feel. Generally speaking, the swing happens anywhere between 1:1 (no swing at all) and 3:1 (where the first note is a dotted eighth note and the second one a sixteenth note); the most common ratio is around 2:1, which sounds like a triplet of eighth notes where the first one is a quarter note and the second one is an eighth note.

USTA implements *Swing* by delaying the second note in each pair. Delay amount is indicated in percentage of note length. For instance, a swing setting of 50% means that the second note will be delayed by half of its length, making the first one 50% longer, which translates in a swing ratio of 3:1; a swing setting around 33% or 34% translates roughly into a swing ratio of 2:1 and so on. The maximum setting allowed by USTA is 75%, which provides an extreme ratio of 7:1 – this goes way beyond the conventional swing styles, and it can be used for more experimental rhythmic structures.

The *Swing* parameter operates between couples of *units* (see above, §2.2). If two stages have length of 1 unit each, the effect will be noticeable. If they have length of 2, the swing theoretically happens within the stages, so it will not be audible. If they have length of 3, the swing happens between the last unit of the first stage and the first unit of the second stage, and so on. In other words, the swing can be heard only if the length of at least one out of two stages has an odd number of units.

If the selected pattern has an even number of stages, the swing pattern will be repeated consistently across different loops; if the pattern has an odd number of stages, the last stage will be read by USTA as the first element of a pair of notes, whose second element will be the first note of the following pattern, or the same pattern if it is looping. This last scenario results in alternating rhythmic variations on the same pattern, because on the second loop all the upbeats and downbeats will be flipped.

8.5 Current Stage Data

In *Performance Mode*, either in *Pattern* or *Song*, the fifth row of the dashboard can be set to display in real time the five channels (*CV A*, *CV B*, *Gate A*, *Gate B*) of the stage that is currently being played by the playhead, instead of the last edited stage.

To activate this option, enter the *Project Menu*, scroll until the `ShowInPLay` item, and select `Yes` (by default, this option is set to `No`, as it could happen that it may slow down USTA).

9 PLAYING IN TUNE

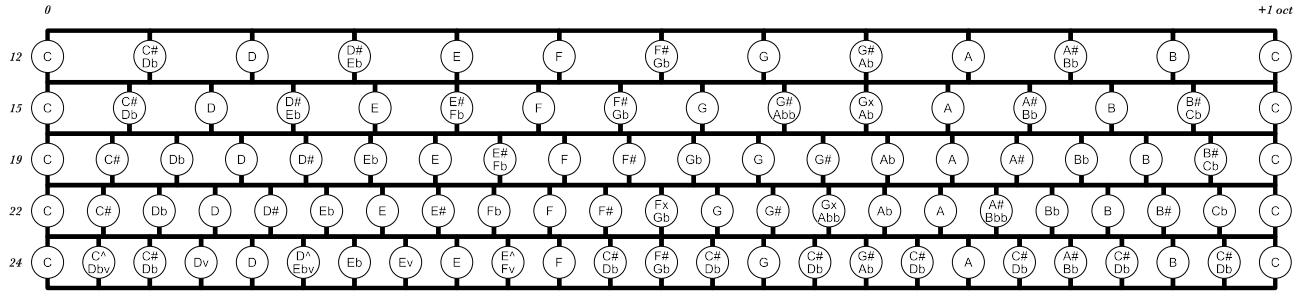


Figure 85: Visual representation of alternative octave divisions.

432		434		436		438		440		442		444		446	
C	A	C	A	C	A	C	A	C	A	C	A	C	A	C	A
32.11	27.00	32.26	27.13	32.41	27.25	32.55	27.38	32.70	27.5	32.85	27.63	33.00	27.75	33.15	27.88
64.22	54.00	64.51	54.25	64.81	54.50	65.11	54.75	65.41	55.00	65.70	55.25	66.00	55.50	66.30	55.75
128.43	108.00	129.03	108.50	129.62	109.00	130.22	109.50	130.81	110.00	131.41	110.50	132.00	111.00	132.60	111.50
256.87	216.00	258.06	217.00	259.25	218.00	260.44	219.00	261.63	220.00	262.81	221.00	264.00	222.00	265.19	223.00
513.74	432.00	516.12	434.00	518.49	436.00	520.87	438.00	523.25	440.00	525.63	442.00	528.00	444.00	530.39	446.00
1027.47	864.00	1032.23	868.00	1036.99	872.00	1041.75	876.00	1046.50	880.00	1051.26	884.00	1056.00	888.00	1060.77	892.00
2054.95	1728.00	2064.46	1736.00	2073.98	1744.00	2083.49	1752.00	2093.00	1760.00	2102.52	1768.00	2112.00	1776.00	2121.55	1784.00

Table 11: C and A frequency chart for different tunings.

9.1 Root & Scale – Dynamic Quantization

In *Pitch* mode, the scale selected, according to the *Root* note, defines how to quantize the outgoing semitones in 1V/oct reference. (In *Raw* mode, the selected scale and root note are ignored, and no quantization is performed.)

To select a *Root* for a track, press its track button, push the *Navigation Encoder* (B.14) once to access the *Track Menu*, and scroll up to reach the `Root` menu item: push the navigation encoder to view the root options, scroll to select the desired root note for your scale, and push again to confirm.

Next, scroll to select the `Scale` menu item: push the navigation encoder again to view the scale options, scroll to select the desired scale, and push the encoder once again to confirm.

It is also possible to choose one of four *Quantization Directions*, used to determine the quantized values for input voltages, first within the selected base scale (e.g. 12EDO), then within the specific scale version (e.g. Major Pentatonic 1):

`Near Up` – finds the nearest match in the selected base scale, starting from the first higher, then checks the first lower, then the second higher, then the second lower and so on until the first match for the selected scale version is found.

`Near Down` – finds the nearest match in the selected base scale, starting from the first lower, then checks the first higher, then the second lower, then the second higher and so on until the first match is found.

`Up` – finds the nearest match in the selected base scale, always increasing the pitch, so all lower semitones are ignored. It checks then the first higher, then the second, then the third and so on until a match for the selected scale version is found.

`Down` – finds the nearest match in the selected base scale, always decreasing the pitch, so all higher semitones are ignored. It checks the first lower, then the second, then the third and so on until a match for the selected scale version is found.

To do that, scroll to select the `Qntz Dir` menu item: push the navigation encoder to view the options, scroll to select the desired quantization direction for your scale, and push again to confirm. All the generated notes will be quantized following these four rules.

Another useful feature is that when the *Variation Range* is active for a stage, the result of the variation is routed to the quantizer, in order to output a note that consistently matches the selected quantization settings.

See the *Scales* section below (§12) for a reference table of the scales available.

Practice the *Dynamic Quantization* settings with this Technique:
Quantization

9.2 Microtonalities

The fact that the equal temperament is the most common in Western music does not mean that it should be the only harmonic framework for your compositions.

For more experimental composers, USTA allows choosing octave divisions for microtonal music different from the standardized 12-EDO (Equal Division of

Octave). Four of them are available: 15-EDO, 19-EDO, 22-EDO, and 24-EDO.

To select an alternate octave division, open the Track Menu and scroll until the `TonesPerOct` item. The default option is `12EDO`, which is the standard Western equal temperament. When `12EDO` is selected, the *Scale* and *Root* options (described in the *Scale&Root* section, above) determine the quantization.

When another option is selected, scale and root are obtained from the menu items below, according the number of tones per octave desired. For example, when `TonesPerOct` is set to `19EDO`, scale and root are determined by the options set in `Root 19` and `Scale 19`, respectively: `Root 15`, `Scale 15`, `Root 19`, `Scale 19`, `Root 22`, `Scale 22`, `Root 24`, `Scale 24`.

A wide variety of preset scales are available (see the Scales section), but it is also possible for the user to define up to four custom scales per each octave division. The settings specified for the *Quantization Direction* will apply to every octave division selected.

Whenever an octave division higher than 12 is selected, the visual feedback system of the *Stage Arc* (A.2) will change accordingly. After the encoder has reached the 11th tone, and the 11th LED of the Stage Arc has lit up, the following intervals will be displayed by the leftmost LEDs tuning off, until the last one (15th, 19th, 22nd or 24th) is reached; then, the following octave will begin from scratch with another LED lighting up on the rightmost side. When the Coarse button is held down, the encoder will always shift the stage value by 1 octave per click and holding the *Fine* button will always shift the pitch by Cents.

Please note that higher EDOs will result in a smaller octave range, because the total number of notes that can be performed by USTA stays the same. (For example, if a 24EDO is selected, USTA's range will be 5 octaves instead of 10).

External pitch shifting is still available, however the 1V/oct standard is no longer in use. In these microtonal modes, 1/12v corresponds to a single interval, no matter the octave division: in other words, if a standard V/Oct keyboard is used to perform Pitch Shift, a semitone above will shift the melody of 1 microtonal interval. If the 24 EDO is selected, two octaves on a keyboard will correspond to one octave on USTA.

This 1/12V-per-interval is available also in Composition Mode via external controller: each key of a standard V/Oct keyboard will input a microtone.

Please note that if you change your Octave Division after you composed a melody in 12EDO, the input melody will still be available, but scaled to the first 12 tones of the new scale. To hear your original melody again, simply return to the standard 12EDO.

9.3 Custom Scales

Beside the built-in scales it is possible to create other custom scales for quantization: up to four scales can be created per each octave division (12, 15, 19, 22 and 24 intervals), for a total of 20 user scales.

This operation must be performed on a computer, as it requires the creation of a .csv file which must be then loaded into the SD card. To create the .csv file we recommend to use a flexible text editor, such as Atom: other editors or word processors might add some unwanted data to the file, such as text formatting, which may make it unreadable for USTA.

The .csv file must be made of four lines, one per each custom scale. Each line must have a succession of digits, which can be only 0 and 1 and must be separated by a semicolon (`;`). The digit represents the semitones, which can be either active (1) or bypassed (0) during the quantization process. The digits will be 12 for the 12-tone scales, 15 for the 15-tone scale and so on, up to the 24-tone scale. No space must be used between digits and semicolons, and no semicolon must be placed after the last digit.

Here is an example of how a .csv file for four custom 12-tone scales will look like: the first scale is chromatic, the second major natural, the third minor natural, the fourth chromatic.

```
1;1;1;1;1;1;1;1;1;1;1;1
1;0;1;0;1;1;0;1;0;1;0;1
1;0;1;1;0;1;0;1;1;0;1;0
1;1;1;1;1;1;1;1;1;1;1;1
```

Please note that the four lines must be completed in full for the file to be properly read by USTA: if you need to create less than four custom scale, consider filling the other lines with e.g. chromatic scales (all 1s).

If an interval is missing at the end of each line, it will be replaced with a 0. Values after the last one will be ignored.

Once your file is ready, you must save it as follows:

Number of tones per octave	File name
12	12scales.csv
15	15scales.csv
19	19scales.csv
22	22scales.csv
24	24scales.csv

Table 12: Custom scales file names.

Once the files are correctly saved, copy them to the root of the SD card, and insert the card into the card holder in the back of USTA. On startup, USTA will automatically read them and the scales will be available as `User1`, `User2`, `User3`, and `User4` under each of the five octave divisions' scale section, at the end of the track menu. To unmount the SD card and connect it to your computer,

refer to the procedure listed in following sections about *Project Management* or *Firmware Update* (§§10.3-10.4).

If no .csv file is found into the SD card, or if the file loaded is not readable by USTA, the four user scales will be replaced by chromatic scales, which is also the condition of USTA at its first boot.

9.4 Set the Reference Note

Traditionally 0V –in the V/oct tracking reference– is intended to be a C. Since the V/oct standard is not absolute but relative, and since that voltage reference truly depends on the tuning frequency at 0V of the oscillator you are using, it seems that some people prefer the C at 0V, while some others prefer the A: in the end, it's just personal taste, even here at Frap Tools we are divided. We decided for this reason to put a setting item in the project menu to define that per each project. This is only a visual reference, but it is fair that everyone can maintain their preferred way of seeing things, to ease the creation and arrangement. The only thing that changes selecting the one or the other is the visualization on the display of the name value of the last edited stage and the current playing stage in a track that uses a *Pitch* mode for their *CV A* or *CV B*.

In order to set the reference note, open the Project Menu, scroll to the item `0v is` and select `C` or `A`.

9.5 Custom Temperaments

USTA by default divides an octave in equal temperaments, which means that the distance between each interval is the same across the whole scale, regardless of how many intervals per octave are chosen. It is possible, however, to create custom tunings to fine-tune certain scale intervals and obtain, for instance, more "perfect" intervals such as thirds or fifths.

The procedure is similar to the one just described for custom scales: a .csv file is required for each of the octave divisions (12, 15, 19, 22, 24 intervals) and the custom scales will be available as option in the `TonesPerOct` menu item.

By default, no file is loaded into the SD card, and these menu items will be a copy of their respective EDO temperaments. The .csv file must be made of four lines, one per each custom temperament. Each line must have a succession of digits, which represent the tuning offset from EDO, expressed in cents, and whose values can be anything from −50 to +50. They must be separated by a semicolon (`;`) and there must be 12 for the 12-tone scales, 15 for the 15-tone scale and so on, up to the 24-tone scale. No space must be used between digits and semicolons, and no semicolon must be placed after the last digit.

Please note that the custom temperaments will not be audible in *Composition* mode, where they are replaced by the default EDO. Temperaments are audible when USTA is playing the sequence either in *Edit* or *Playback* mode.

Here is an example of how a single line of the .csv file for a custom 12-tone temperament will look like.

0;0;5;0;-7;0;0;11;0;-2;0;0

In this example, the major second is raised by 5 cents, the major third is lowered by 7, the fifth is raised by 11 and the sixth is lowered by two.

Please note that the four lines must be completed in full for the file to be properly read by USTA: if you need to create less than four custom temperaments, consider filling the other lines with 0s.

If a value is missing at the end of each line, it will be replaced with a 0. Values after the last one will be ignored. Once your file is ready, you must save it as follows:

Number of tones per octave	File name
12	12temper.csv
15	15temper.csv
19	19temper.csv
22	22temper.csv
24	24temper.csv

Table 13: Custom temperaments file names.

Once the files are correctly saved, copy them to the SD card and insert the card into the card holder in the back of USTA. On startup, USTA will automatically read them and the temperaments will be available as `12User1`, `12User2`, `12User3`, `12User4`, `15User1`, `15User2` and so on, within the `TonesPerOct` item in the track menu. To unmount the SD card and connect it to your computer, refer to the procedure listed in following sections about about *Project Management* or *Firmware Update* (§§10.3-10.4).

9.5.1 Absolute or Relative Temperaments

After defining your tuning, it is possible to choose whether quantization is calculated by USTA on the basis of the 0V note (absolute) or in relation to the Root note selected for the respective octave division (relative).

The reason for doing so is that one may need a given temperament for different compositions in different keys. To set this option, enter the Project Menu, scroll until the `Temperament` item and select `Relative` or `Absolute` as default behavior. Again, this setting affects USTA as a whole, and does not belong to a specific project.

Important note! The .csv files created for Scales and Temperaments are global parameters, just like the trimming file. It means that they are not saved in each project, but they are recalled every time. If you need a specific scale or temperament for a project, make sure not to replace it on the .csv file!

9.6 LED Pitch Tables

semitones	0V is C	C	C#	D	D#	E	F	F#	G	G#	A	A#	B
	0V is A	A	A#	B	C	C#	D	D#	E	F	F#	G	G#
	Voltage	0V	1/12V ~0.083V	2/12V ~0.166V	3/12V 0.25V	4/12V ~0.333V	5/12V ~0.416V	6/12V 0.5V	7/12V ~0.583V	8/12V ~0.666	9/12V 0.75V	10/12V ~0.833V	11/12V ~0.916V
octaves	Octave	+0	+1	+2	+3	+4	+5	+6	+7	+8	+9	+10	
	Voltage	0V	1V	2V	3V	4V	5V	6V	7V	8V	0V	0V	

Table 14: 12-EDO LED chart.

semitones	0V is C	C	C#	D	D#	E	E#	F	F#	G	G#	Ab	A
	0V is A	A	A#	B	B#	C	C#	D	D#	E	E#	F	F#
	Voltage	0V	1/15V ~0.066V	2/15V ~0.133V	3/15V 0.2V	4/15V ~0.266V	5/15V ~0.333V	6/15V 0.4V	7/15V ~0.466V	8/15V ~0.533	9/15V 0.6V	10/15V ~0.666V	11/15V ~0.733V
	0V is C	A#	B	B#									
	0V is A	G	G#	Ab									
	Voltage	12/15V ~0.8V	13/15V ~0.866V	14/15V ~0.933V									
octaves	Octave	+0	+1	+2	+3	+4	+5	+6	+7	+8			
	Voltage	0V	1V	2V	3V	4V	5V	6V	7V	8V			

Table 15: 15-EDO LED chart.

semitones	0V is C	C	C#	Db	D	D#	Eb	E	E#	F	F#	Gb	G
	0V is A	A	A#	Bb	B	B#	C	C#	Db	D	D#	Eb	E
	Voltage	0V	1/19V ~0.052V	2/19V ~0.105V	3/19V ~0.157V	4/19V ~0.210V	5/19V ~0.263V	6/19V ~0.315V	7/19V ~0.368V	8/19V ~0.421	9/19V~ 0.473V	10/19V ~0.526V	11/19V ~0.578V
	0V is C	G#	Ab	A	A#	Bb	B	B#					
	0V is A	E#	F	F#	Gb	G	G#	Ab					
	Voltage	12/19V ~0.631V	13/19V 0.684V	14/19V ~0.736V	15/19V ~0.789V	16/19V ~0.842V	17/19V ~0.894V	18/19V ~0.947V					
octaves	Octave	+0	+1	+2	+3	+4	+5	+6					
	Voltage	0V	1V	2V	3V	4V	5V	6V					

Table 16: 19-EDO LED chart.

semitones	0V is C	C	C#	Db	D	D#	Eb	E	E#	Fb	F	F#	Gb
	0V is A	A	A#	Bb	B	B#	Cb	C	C#	Db	D	D#	Eb
	Voltage	0V	1/22V ~0.045V	2/22V ~0.090V	3/22V ~0.136V	4/22V ~0.181V	5/22V ~0.227V	6/22V ~0.272V	7/22V ~0.318V	8/22V ~0.363	9/22V ~0.409V	10/22V ~0.454V	11/22V 0.5V
	0V is C	G	G#	Abb	Ab	A	A#	Bb	B	B#	Cb		
	0V is A	E	E#	Fb	F	F#	Gb	G	G#	Abb	Ab		
	Voltage	12/22V ~0.545V	13/22V ~0.590V	14/22V ~0.636V	15/22V ~0.681V	16/22V ~0.727V	17/22V ~0.772V	18/22V ~0.818V	19/22V ~0.863V	20/22V ~0.909V	21/22V ~0.954V		
octaves	Octave	+0	+1	+2	+3	+4	+5						
	Voltage	0V	1V	2V	3V	4V	5V						

Table 17: 22-EDO LED chart.

semitones	0V is C	C	C^	C#	Dv	D	D^	D#	Ev	E	E^	F	F^
	0V is A	A	A^	A#	Bv	B	B^	C	C^	C#	Dv	D	D^
	Voltage	0V	1/24V ~0.041V	2/24V ~0.083V	3/24V 0.125V	4/24V ~0.166V	5/24V ~0.208V	6/24V 0.25V	7/24V ~0.291V	8/24V ~0.333	9/24V 0.375V	10/24V ~0.416V	11/24V ~0.458V
	0V is C	F#	Gv	G	G^	G#	Av	A	A^	A#	Bv	B	B^
	0V is A	D#	Ev	E	E^	F	F^	F#	Gv	G	G^	G#	Av
	Voltage	12/24V 0.5V	13/24V ~0.541V	14/24V ~0.583V	15/24V 0.625V	16/24V ~0.666V	17/24V ~0.708V	18/24V 0.75V	19/24V ~0.791V	20/24V ~0.833V	21/24V 0.875V	22/24V ~0.916V	23/24V ~0.958V
octaves	Octave	+0	+1	+2	+3	+4							
	Voltage	0V	1V	2V	3V	4V							

Table 18: 24-EDO LED chart.

10 ADDITIONAL MAINTENANCE

10.1 Remove / Insert the SD Card

For some operations it is required to remove and then reinsert the SD card. We won't copy/paste the same process here every time so here's how to do that.

10.1.1 Remove the SD Card

(a) Turn off the module.
(b) Unmount it from your case removing all the screws.
(c) Unplug the module power cable.
(d) Take off the SD card from the back of the module by pushing it (it's a push-push mechanism).

10.1.2 Insert the SD Card

(e) Safe remove the SD card from the computer.
(f) Put the SD card back in place onto the module: if you are looking at the back of the module, the SD card contacts should look at you. Insert the SD card into the SD card holder and push to retain it.
(g) Plug the module power cable.
(h) Remount it safely in your case using all the screws.
(i) You can now turn on the module.

10.2 Analog Trimming

The trimming routine is an alternative way to start the machine, to perform, set, adjust digital trimming on any of the CV output or input.

Each unit's CV inputs and outputs are software calibrated to guarantee the most precise response. For this purpose, it is highly recommended to make a backup copy of your USTA's trimming file. To do so, unmount the SD card (see instructions above), connect it to your computer and copy the file labeled trim.cfg, then place the SD card back into the module (see instructions above). Should you lose somehow the file on the SD card, USTA will automatically create a new trimming file with default values, which may cause imprecise tracking.

To access the *Trimming* turn off the USTA, hold the *Play/Pause* button (B.13), and turn on the USTA again.

The trimming menu will show you 32 settings for the outputs and 4 settings for the inputs. Each output or input needs to be set for −5V, 0V, +5V, and +10V. Scroll through elements with the navigation encoder, push it to start to modify the value, and push again to confirm it.

For best results, we strongly suggest warming up the USTA at least for 30 minutes inside the case where it will be used before performing any sort of trimming.

To perform the trimming on the outputs a multimeter is needed. Consider also the precision of the multimeter you are using. The USTA outputs are based on an octal 16-bit DAC, with 65'536 steps: the maximum output without trimming is from around −5.5V up to around +10.5V, which translates on 16V of range and an amplitude difference between each step of around 0.244 mV.

To perform the trimming on the inputs, a precise voltage generator is needed, with the capability to generate from at least −5.5V up to +10.5V.

To save what you did, push the play button.

When you are done, you can turn off and normally turn on again the USTA.

10.3 Project Management and Backups

Projects can be backed up on your computer, and it is strongly suggested to back them up. To do so:

Turn off and unplug your case.

Remove the SD Card (follow instructions above).

Connect the SD card to the computer you want to use for backups.

Copy/paste the project files you want to save from the SD card root to your computer.

The project is now backed up.

Insert the SD Card (follow instructions above).

10.4 Firmware Update

(a) Turn off and unplug your case.
(b) Remove the SD Card (follow instructions above).
(c) Download the firmware file from https://frap.tools/firmwares/ to your computer
(d) Connect the SD card to that computer.
(e) Rename the downloaded file as *USTA.bin* (keep the extension .bin, it is important).
(f) Place the renamed file on the root of the SD card – if there is another USTA.bin, replace it (if you have an older install in there and you don't replace it, you know what happens right?).
(g) Insert the SD Card (follow instructions above) and turn on the module.
(h) Press and hold the navigation encoder until the Project Menu is displayed.
(i) Select `Replace FW` and push the encoder. A window will open, asking for confirmation. By default, the answer is `NO`, to prevent accidental misbehaviors. Rotate the navigation encoder counterclockwise to select `YES`, then push the encoder to launch the upgrade.

The device will automatically reboot and begin the update procedure. You must wait for it to complete the update before using it normally. DO NOT TURN OFF POWER FROM THIS POINT, OR IT WILL **VOID** YOUR MODULE'S **WARRANTY**!

11 CHARTS

11.1 PROJECT MENU

Menu voice	Selectable options	Function
New...	—	Creates a new file.
Load...	—	Loads any existent file.
Save	—	Saves the changes made to the current project.
Save As...	—	Saves the current project as a new file.
Delete...	—	Deletes any existent file.
Rename...	—	Renames any existent file.
Aux Target	Reset / Run 1 / Run 2 / Run 3 / Run 4	Selects the behavior of the external gate signal.
Master Track	1 / 2 / 3 / 4	Selects the Master Track.
ShowInPlay	No / Yes	Enable to display in real time the playing stage in PerformanceMode in the dashboard.
0V is	C / A	Selects the Reference Note.
All Edits	From / All	Selects the behavior of the Set All and Shift All buttons.
Temperament	Absolute / Relative	Selects if the User Temperaments are applied to the Reference Note only, or to the root note selected in each track (which may change).
Replace FW	—	Updates the current firmware (when a new firmware file is loaded into the SD card).

11.2 TRACK MENU

Menu voice	Selectable options	Function
Int BPM	from 30 to 300	Select the internal clock speed expressed in BPM, the clock source, and the clock-to-unit ratio, as expressed in the chapters about Tempo Management and Editing Tracks.
ClockSource	Built-In / External	
Ratio	...	
Swing	0 - 50	Sets the Swing ratio.
Track Mode	Pattern / Song	Selects the track mode.
CV A Mode	Pitch / Raw	Sets the CV mode for channel A and B to raw voltages or quantized notes.
CV B Mode	Pitch / Raw	
CV A Range	0/+10V / -/+5V	Defines if the CV range for channel A and B is unipolar or bipolar.
CV B Range	0/+10V / -/+5V	
Root	C - B	Defines the root note and the scale in the default 12EDO system. See also the 12-EDO scales section.
Scale	...	
Quantize	Near Up / Near Down / Up / Down	Selects the dynamic quantization direction.
Gate A % Width	10 - 90	Sets the time a gate stays high when in ratcheting mode (green color), expressed in the % of the time between two consecutive rising edges.
Gate B % Width	10 - 90	
Reset What	Nothing / Stage / Stage&Pa	Select the reset options.
Reset On	Instant / Local / Master	
PitchShiftA	None / CV A / CV B	Sets the Pitch Shift source for CV A and B.
PitchShiftB	None / CV A / CV B	
Root Shift	None / CV A / CV B	Sets the Root Shift source.
StageShift	None / CV A / CV B	Sets the Stage Shift source.
Ptrn Shift	None / CV A / CV B	Sets the Pattern Shift source.
PhaseShift	None / CV A / CV B	Sets the Phase Shift source.
Gate ShiftA	None / CV A / CV B	Sets the Gate Shift source for GATE A and B.
Gate ShiftB	None / CV A / CV B	
Vari Shift	None / CV A / CV B	Sets the Variation Shift source.
TonesPerOct	...	Sets the octave division unit or the custom temperament.
Root 15	...	Select the reset options. Defines the root and scale for each of the other four octave divisions available. See also the following sections about. 15, 19, 22 and 24-EDO scales.
Scale 15	...	
Root 19	...	
Scale 19	...	
Root 22	...	
Scale 22	...	
Root 24	...	
Scale 24	...	

11.3 PROJECT HIERARCHY

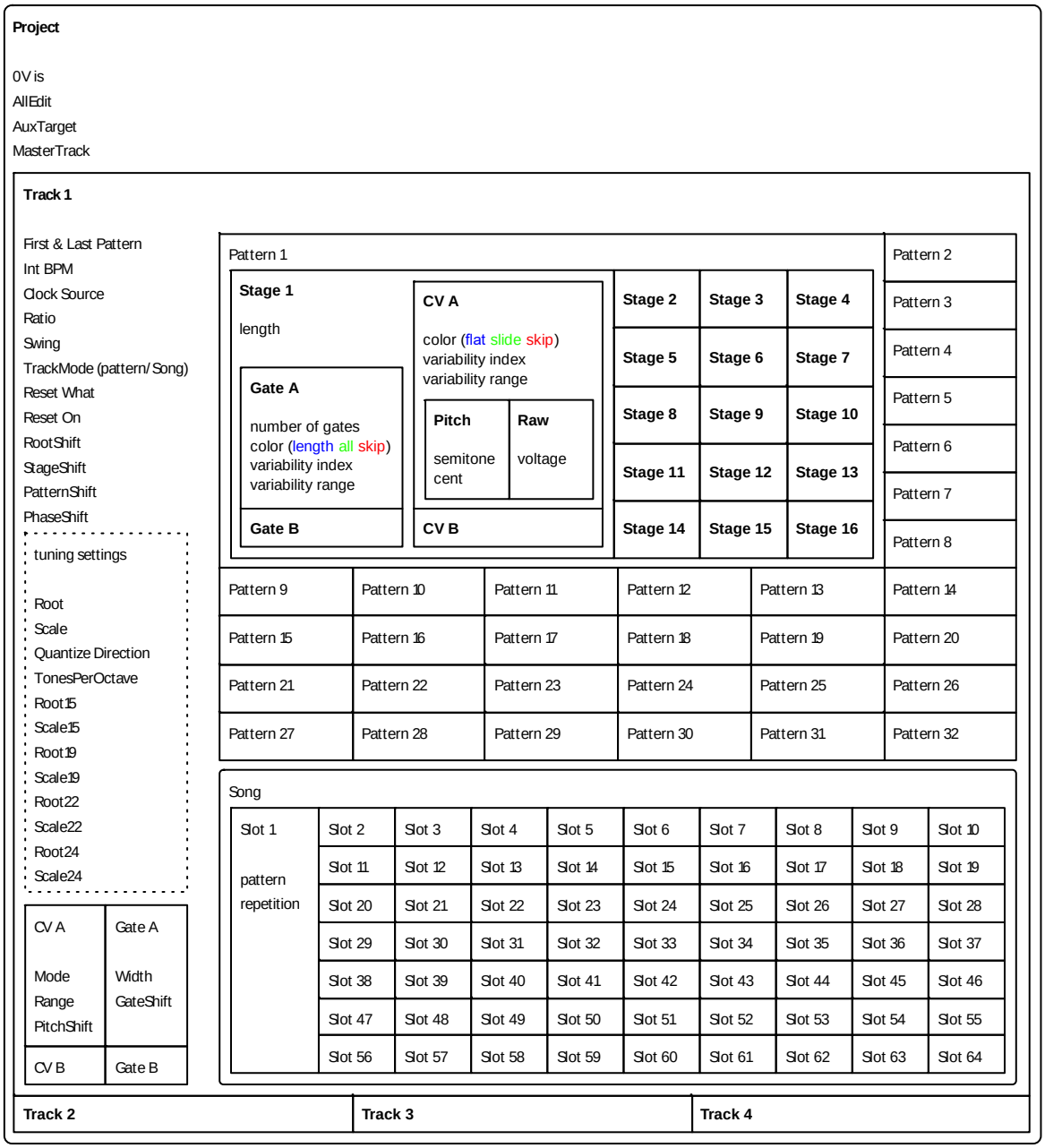


Figure 86: USTA structure.

12 SCALE TABLES

Table 19: 12-EDO and other 12-tone-per-octave temperaments.

		tone	0	1	2	3	4	5	6	7	8	9	10	11
		cents	0	100	200	300	400	500	600	700	800	900	1000	1100
description	t/oct	display												
Chromatic	12	Chroma	1	1	1	1	1	1	1	1	1	1	1	1
Major Pentatonic 1	5	5 Maj 1	1	0	1	0	1	0	0	1	0	1	0	0
Minor Pentatonic 2	5	5 Min 2	1	0	1	0	0	1	0	1	0	0	1	0
Minor Pentatonic 3	5	5 Min 3	1	0	0	1	0	1	0	0	1	0	1	0
Major Pentatonic 2	5	5 Maj 2	1	0	1	0	0	1	0	1	0	1	0	0
Minor Pentatonic 1	5	5 Min 1	1	0	0	1	0	1	0	1	0	0	1	0
Dominant Pentatonic	5	5 Domin	1	0	1	0	1	0	0	1	0	0	1	0
Whole Tone	6	6 Whole	1	0	1	0	1	0	1	0	1	0	1	0
Augmented	6	6 Augmt	1	0	0	1	1	0	0	1	1	0	0	1
Prometheus	6	6 Mystc	1	0	1	0	1	0	1	0	0	1	1	0
Hexatonic Blues	6	6 Blues	1	0	0	1	0	1	1	1	0	0	1	0
Ionian	7	Ionian	1	0	1	0	1	1	0	1	0	1	0	1
Dorian	7	Dorian	1	0	1	1	0	1	0	1	0	1	1	0
Phrygian	7	Phrygia	1	1	0	1	0	1	0	1	1	0	1	0
Lydian	7	Lydian	1	0	1	0	1	0	1	1	0	1	0	1
MixoLydian	7	Mixo	1	0	1	0	1	1	0	1	0	1	1	0
Aeolian	7	Aeolian	1	0	1	1	0	1	0	1	1	0	1	0
Locrian	7	Locrian	1	1	0	1	0	1	1	0	1	0	1	0
Harmonic Minor — Aeolian #7, m Harmonic Modes	7	mHarmon	1	0	1	1	0	1	0	1	1	0	0	1
Locrian #6	7	Locri#6	1	1	0	1	0	1	1	0	0	1	1	0
Ionian #5	7	Ionia#5	1	0	1	0	1	1	0	0	1	1	0	1
Dorian #4	7	Doria#4	1	0	1	1	0	0	1	1	0	1	1	0
Phrygian Dominant	7	PhryDom	1	1	0	0	1	1	0	1	1	0	1	0
Lydian #2	7	Lydia#2	1	0	0	1	1	0	1	1	0	1	0	1
Ultralocrian	7	uLocria	1	1	0	1	1	0	1	0	1	1	0	0
Melodic Minor, Jazz Minor	7	mMelodi	1	0	1	1	0	1	0	1	0	1	0	1
Dorian b9	7	Doriab9	1	1	0	1	0	1	0	1	0	1	1	0
Lydian Augmented	7	LydiAug	1	0	1	0	1	0	1	0	1	1	0	1
Lydian Dominant	7	LydiDom	1	0	1	0	1	0	1	1	0	1	1	0
Mixolydian b6	7	Mixo b6	1	0	1	0	1	1	0	1	1	0	1	0
Semilocrian, Aeolian b5	7	sLocria	1	0	1	1	0	1	1	0	1	0	1	0
Superlocrian	7	SuLocri	1	1	0	1	1	0	1	0	1	0	1	0
Double Harmonic Major	7	dHarmMa	1	1	0	0	1	1	0	1	1	0	0	1
Lydian #2 #6	7	Lyd#2#6	1	0	0	1	1	0	1	1	0	0	1	1
UltraPhrygian, Ultralocrian natural 5	7	uPhrygi	1	1	0	1	1	0	0	1	1	1	0	0
Double Harmonic Minor, Hungarian Minor	7	dHarmMi	1	0	1	1	0	0	1	1	1	0	0	1
Mixolydian b2 b5, Oriental	7	Mixb2b5	1	1	0	0	1	1	1	0	0	1	1	0
Ionian Augmented #2	7	IoAug#2	1	0	0	1	1	1	0	0	1	1	0	1
Locrian bb3 bb7	7	Locbb37	1	1	1	0	0	1	1	0	1	1	0	0
Hungarian Major	7	Hungari	1	0	0	1	1	0	1	1	0	1	1	0
Superlocrian bb6 bb7	7	SuLbb67	1	1	0	1	1	0	1	1	0	1	0	0
Harmonic Minor b5	7	mHarmb5	1	0	1	1	0	1	1	0	1	0	0	1
Superlocrian #6	7	SuLoc#6	1	1	0	1	1	0	1	0	0	1	1	0
Melodic Minor #5, Jazz Minor #5	7	mMelo#5	1	0	1	1	0	1	0	0	1	1	0	1
Dorian b9 #11	7	Dor#4b9	1	1	0	1	0	0	1	1	0	1	1	0
Lydian Augmented #3	7	LydAu#3	1	0	1	0	0	1	1	0	1	1	0	1
Harmonic Major, Ionian b6	7	MHarmon	1	0	1	0	1	1	0	1	1	0	0	1
Dorian b5	7	Doriab5	1	0	1	1	0	1	1	0	0	1	1	0
Phrygian b4	7	Phrygb4	1	1	0	1	1	0	0	1	1	0	1	0
Lydian b3	7	Lydiab3	1	0	1	1	0	0	1	1	0	1	0	1
Mixolydian b9	7	Mixo b9	1	1	0	0	1	1	0	1	0	1	1	0
Lydian Augmented #2	7	LydAu#2	1	0	0	1	1	0	1	0	1	1	0	1
Locrian bb7	7	Loc bb7	1	1	0	1	0	1	1	0	1	1	0	0
Neapolitan Major	7	MNeapol	1	1	0	1	0	1	0	1	0	1	0	1
Leading Whole-Tone	7	LeaWhol	1	0	1	0	1	0	1	0	1	0	1	1
Lydian Augmented Dominant	7	LydAuDo	1	0	1	0	1	0	1	0	1	1	1	0
Lydian Dominant b6	7	LydDob6	1	0	1	0	1	0	1	1	1	0	1	0
Major Locrian	7	MLocria	1	0	1	0	1	1	1	0	1	0	1	0
Semilocrian b4	7	sLoc b4	1	0	1	1	1	0	1	0	1	0	1	0
Superlocrian bb3	7	SuLobb3	1	1	1	0	1	0	1	0	1	0	1	0
Neapolitan Minor	7	mNeapol	1	1	0	1	0	1	0	1	1	0	0	1
Lydian #6	7	Lydia#6	1	0	1	0	1	0	1	1	0	0	1	1
Mixolydian Augmented	7	MixoAug	1	0	1	0	1	1	0	0	1	1	1	0
Aeolian #4, Hungarian Gipsy	7	Aeoli#4	1	0	1	1	0	0	1	1	1	0	1	0
Locrian Dominant	7	Loc Dom	1	1	0	0	1	1	1	0	1	0	1	0
Ionian #2	7	Ionia#2	1	0	0	1	1	1	0	1	0	1	0	1
Ultralocrian bb3	7	uLocbb3	1	1	1	0	1	0	1	0	1	1	0	0
Chromatic Hypolydian	7	hc Lydi	1	1	0	0	1	0	1	1	1	0	0	1

Chromatic Hypophrygian	7	hc Phri	1	0	0	1	0	1	1	1	0	0	1	1
Chromatic Hypodorian	7	hc Dori	1	0	1	1	1	0	0	1	1	1	0	0
Chromatic Mixolydian	7	hc Mixo	1	1	1	0	0	1	1	1	0	0	1	0
Chromatic Lydian	7	c Lydia	1	1	0	0	1	1	1	0	0	1	0	1
Chromatic Phrygian	7	c Phryg	1	0	0	1	1	1	0	0	1	0	1	1
Chromatic Dorian	7	c Doria	1	1	1	0	0	1	0	1	1	1	0	0
Chromatic Hypophrygian inverse	7	hciPhry	1	1	1	0	0	1	1	1	0	1	0	0
Chromatic Hypolydian inverse	7	hciLydi	1	1	0	0	1	1	1	0	1	0	0	1
Chromatic Dorian inverse	7	hciDori	1	0	0	1	1	1	0	1	0	0	1	1
Chromatic Phrygian inverse	7	hciPhri	1	1	1	0	1	0	0	1	1	1	0	0
Chromatic Lydian inverse	7	ci Lydi	1	1	0	1	0	0	1	1	1	0	0	1
Chromatic Mixolydian inverse	7	ci Mixo	1	0	1	0	0	1	1	1	0	0	1	1
Chromatic Hypodorian inverse	7	ci Dori	1	0	0	1	1	1	0	0	1	1	1	0
BeBop Major	8	bbp Maj	1	0	1	0	1	1	0	1	1	1	0	1
BeBop Dominant	8	bbp Dom	1	0	1	0	1	1	0	1	0	1	1	1
BeBop Dorian	8	bbp Dor	1	0	1	1	1	1	0	1	0	1	1	0
BeBop Dorian Alternative	8	bbp DoA	1	0	1	1	0	1	0	1	0	1	1	1
BeBop Minor	8	bbp Min	1	0	1	1	0	1	0	1	1	1	0	1
BeBop Locrian	8	bbp Loc	1	1	0	1	0	1	1	1	1	0	1	0
BeBop Harmonic Minor — Natural Minor	8	bbp mHa	1	0	1	1	0	1	0	1	1	0	1	1
Diminished, Whole-Half	8	Diminis	1	0	1	1	0	1	1	0	1	1	0	1
Octatonic, Half-Whole	8	Octaton	1	1	0	1	1	0	1	1	0	1	1	0
	12	User 1	1	1	1	1	1	1	1	1	1	1	1	1
	12	User 2	1	1	1	1	1	1	1	1	1	1	1	1
	12	User 3	1	1	1	1	1	1	1	1	1	1	1	1
	12	User 4	1	1	1	1	1	1	1	1	1	1	1	1

Table 20: 15-EDO and other 15-tone-per-octave temperaments.

		tone	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14
		cents	0	80	160	240	320	400	480	560	640	720	800	880	960	1040	1120
description	t/oct	display															
Chromatic	15	Chroma	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
Blackwood	6	6 Blckw	1	0	0	0	1	1	0	0	0	1	1	0	0	0	1
15 Tone Miller Porcupine 7	7	7 Millr	1	0	1	0	1	0	1	0	0	1	0	1	0	1	0
15 Tone Miller Porcupine 7 Major	7	7 MillM	1	0	0	1	0	1	0	1	0	1	0	1	0	1	0
Miller's Kusiro	8	8 Kusir	1	1	0	0	1	0	1	0	0	1	1	1	0	1	0
15 Tone Major, Jones's Porcupine 8	8	8 JoneM	1	0	1	0	1	1	0	1	0	1	0	1	0	1	0
Rempt's Andal	9	9 Rempt	1	0	1	0	1	1	0	1	0	1	1	0	1	0	1
Blackwood	10	10Blckw	1	1	0	1	1	0	1	1	0	1	1	0	1	1	0
15 Tone Major-Minor Mix	10	10MmMix	1	0	0	1	1	1	1	0	0	1	1	1	0	1	1
12 Tone Chromatic	12	12Chrom	1	1	0	1	1	1	1	0	1	1	1	1	1	0	1
	15	User 1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
	15	User 2	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
	15	User 3	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
	15	User 4	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1

Table 21: 19-EDO and other 19-tone-per-octave temperaments.

		tone	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18
		cents	0,0	63,2	126,3	189,5	252,6	315,8	378,9	442,1	505,3	568,4	631,6	694,7	757,9	821,1	884,2	947,4	1010,5	1073,7	1136,8
description	t/oct	display																			
Chromatic	19	Chroma	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
Four out of 19	4	Four	1	0	0	0	0	1	0	0	0	1	0	0	0	0	1	0	0	0	0
Five out of 19	5	Five	1	0	0	0	1	0	0	0	1	0	0	0	1	0	0	0	1	0	0
Quasi-Equal Pentatonic	5	qPenta	1	0	0	0	1	0	0	0	1	0	0	1	0	0	0	1	0	0	0
Yasser's Hexad	6	6 Yassr	1	0	0	1	0	0	1	0	0	1	0	0	1	0	0	1	0	0	0
Oljare Diminished	7	7 Olj D	1	0	0	0	1	1	0	0	0	1	1	0	0	0	1	1	0	0	0
Oljare Diminished	7	7 Olj A	1	0	0	0	0	1	1	0	0	0	0	1	1	1	0	0	0	0	1
Oljare Octatonic	8	8 Oljar	1	0	1	0	1	0	0	1	0	1	0	1	0	0	1	0	1	0	0
Oljare Pentaenharmonic	9	9 PenEn	1	1	0	0	1	1	0	0	1	0	0	1	1	0	0	1	1	0	0
Negri's Ten Plus Nine	10	10 Negr	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1
Keenan eleven out of 19	11	11 Keen	1	0	0	1	1	1	0	0	1	1	0	0	1	1	1	0	0	1	1
Krantz eleven out of 19	11	11 Kran	1	0	1	0	1	0	1	0	1	0	1	1	0	1	1	0	1	0	1
McLaren eleven out of 19	11	11 McLa	1	0	0	1	1	1	0	0	1	1	1	1	1	0	0	1	1	0	0
Gould Eleven out of 19	11	11 Goul	1	0	1	0	1	1	0	1	0	1	0	1	1	0	1	0	1	0	1
Meantone Chromat (1/3 comma)	12	MeanChr	1	1	0	1	0	1	1	0	1	1	0	1	1	0	1	0	1	1	0
Genus Diatonico-Chromaticum	12	GenuChr	1	0	1	1	0	1	1	0	1	1	0	1	0	1	1	0	1	1	0
Yasser's Supradiatonic	12	SupraDi	1	0	1	1	0	1	1	0	1	0	1	1	0	1	1	0	1	1	0
Mandelbaum's Eight out of 19	8	8 Mande	1	0	1	0	0	1	0	1	0	1	0	0	1	0	1	0	1	0	0
Mandelbaum's Nine out of 19	9	9 Mande	1	0	1	0	1	0	0	1	0	1	0	1	0	1	0	1	0	1	0
Mandelbaum's Ten out of 19	10	10Mande	1	0	1	0	1	1	0	1	0	1	0	1	0	1	0	1	0	1	0
Mandelbaum's Eleven out of 19	11	11Mande	1	0	1	1	0	1	0	1	0	1	1	0	1	0	1	0	1	1	0
Mandelbaum's Twelve out of 19	12	12Mande	1	1	0	1	1	0	1	0	1	1	0	1	1	0	1	1	0	1	0
Mandelbaum's Thirteen out of 19	13	13Mande	1	0	1	1	0	1	1	0	1	1	0	1	1	0	1	1	0	1	1
Mandelbaum's Fourteen out of 19	14	14Mande	1	1	0	1	1	1	0	1	1	1	0	1	1	1	0	1	1	0	1
	19	User 1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
	19	User 2	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
	19	User 3	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
	19	User 4	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1

Table 22: 22-EDO and other 22-tone-per-octave temperaments.

		tone	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21
		cents	0,0	54,5	109,1	163,6	218,2	272,7	327,3	381,8	436,4	490,9	545,5	600,0	654,5	709,1	763,6	818,2	872,7	927,3	981,8	1036,4	1090,9	1145,5
description	**t/oct**	**display**																						
Chromatic	15	Chroma	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
Twelve-tone Chromatic (1/3-comma positive)	12	12Chrom	1	0	0	1	1	1	0	0	1	1	0	0	1	1	0	0	1	1	1	0	0	1
22 tone Major	7	Major	1	0	0	0	1	0	0	0	1	1	0	0	0	1	0	0	0	1	0	0	0	1
22 tone Melodic Minor	7	mMelodi	1	0	0	0	1	0	1	0	0	1	0	0	0	1	0	0	1	0	0	0	1	0
22 tone Harmonic Minor	7	mHarmon	1	0	0	0	1	0	1	0	0	1	0	0	0	1	0	1	0	0	0	0	1	0
22 tone Harmonic Major	7	MHarmon	1	0	0	0	1	0	0	1	0	1	0	0	0	1	0	1	0	0	0	0	1	0
22 tone Astrology-10	10	Astrolo	1	0	0	1	1	0	0	1	0	0	1	1	0	0	1	1	0	0	1	0	0	1
22 tone Doublewide-10	10	DblWi10	1	1	0	0	0	1	1	0	0	0	1	1	1	0	0	0	1	1	0	0	0	1
22 tone Doublewide-14	14	DblWi14	1	1	1	0	0	1	1	1	0	0	1	1	1	1	0	0	1	1	1	0	0	1
22 tone Fleetwood-14	14	Fltwo14	1	1	1	0	0	1	1	1	1	0	0	1	1	1	0	0	1	1	1	1	0	0
22 tone Hedgehog-6	6	Hedge 6	1	0	0	0	0	1	0	0	1	0	0	1	0	0	0	0	1	0	0	1	0	0
22 tone Hedgehog-8	8	Hedge 8	1	0	1	0	0	1	0	0	1	0	0	1	0	1	0	0	1	0	0	1	0	0
22 tone Hedgehog-14	14	Hedge14	1	0	1	1	0	1	0	1	1	0	1	1	0	1	1	0	1	0	1	1	0	1
22 tone Jubilee-12	12	Jubil12	1	0	1	0	1	1	0	1	0	1	0	1	0	1	0	1	1	0	1	0	1	0
22 tone Pajara-12	12	Pajar12	1	0	1	0	1	0	1	1	0	1	0	1	0	1	0	1	0	1	1	0	1	0
22 tone Supra-5, Septimal Minor Pentatonic	5	Supra 5	1	0	0	0	0	1	0	0	0	1	0	0	0	1	0	0	0	0	1	0	0	0
22 tone Supra-7	7	Supra 7	1	0	0	0	1	1	0	0	0	1	0	0	0	1	0	0	0	1	1	0	0	0
22 tone Supra-12	12	Supra12	1	1	0	0	1	1	0	0	1	1	1	0	0	1	1	0	0	1	1	0	0	1
22 tone Urchin-14	14	Supra14	1	0	1	0	1	1	0	1	1	0	1	1	0	1	0	1	1	0	1	1	0	1
22 tone Wilson Pi-Meantone	12	WilsoPi	1	0	0	1	1	0	0	1	1	0	0	1	1	1	0	0	1	1	0	0	1	1
Ionian Porcupine	7	IoniaPo	1	0	0	0	1	0	0	1	0	1	0	0	0	1	0	0	1	0	0	1	0	0
Dorian Porcupine	7	DoriaPo	1	0	0	0	1	0	1	0	0	1	0	0	0	1	0	0	1	0	0	1	0	0
Aeolian Porcupine	7	AeoliPo	1	0	0	1	0	0	1	0	0	1	0	0	0	1	0	1	0	0	1	0	0	0
Major Porcupine	7	MajorPo	1	0	0	0	1	0	0	1	0	0	1	0	0	1	0	0	1	0	0	0	1	0
Major-Minor Porcupine	7	MajMiPo	1	0	0	0	1	0	0	1	0	0	1	0	0	1	0	1	0	0	0	1	0	0
Chameleon Porcupine	7	ChamePo	1	0	0	0	1	0	0	1	0	0	1	0	0	1	0	0	0	1	0	0	1	0
Symmetric Diminished Porcupine	7	SymDimP	1	0	0	1	0	0	1	0	0	0	1	0	1	0	0	0	1	0	0	1	0	0
Porcupine-15	15	Porcu15	1	0	1	1	0	1	1	1	0	1	1	0	1	1	0	1	1	0	1	1	0	1
Elevenplus	12	Eleven+	1	0	1	0	1	0	1	0	1	0	1	0	1	1	1	0	1	0	1	0	1	0
Hexachordal	12	Hexacho	1	0	1	0	1	0	1	0	1	1	0	1	0	1	0	1	0	1	0	1	0	1
Noll Pseudo-diatonic	13	NollPsD	1	0	1	0	1	1	0	1	0	1	0	1	1	0	1	0	1	1	0	1	0	1
Ballooning Rushes	7	Balloon	1	0	0	1	1	0	0	0	0	0	1	0	1	0	0	0	0	0	1	1	0	0
Crushed Oranges	8	CrushOr	1	1	0	0	1	0	0	0	0	1	1	0	0	1	0	0	0	0	1	1	0	0
Kathartic Parts	7	Kathart	1	0	1	1	0	0	0	0	0	0	1	0	1	1	0	0	0	0	0	0	1	0
Rivetting Reds	9	Rivetti	1	1	0	0	0	1	1	0	0	0	1	0	1	1	0	0	0	1	1	0	0	0
Rodentalia	9	Rodenta	1	0	0	0	1	1	0	0	1	1	0	0	0	1	1	0	0	1	1	0	0	0
Rezsutek's Percussion Scale	9	Rezsute	1	0	1	0	0	1	0	1	0	0	1	0	1	0	0	1	0	1	0	0	1	0
22 tone Magic-7	7	Magic 7	1	0	0	0	0	0	1	1	0	0	0	0	0	1	1	1	0	0	0	0	0	1
22 tone Magic-10	10	Magic10	1	0	0	0	0	1	1	1	1	0	0	0	0	1	1	1	0	0	0	0	1	1
22 tone Magic-13	13	Magic13	1	0	0	0	1	1	1	1	0	0	0	1	1	1	1	0	0	0	1	1	1	1
22 tone Orwell-5	5	Orwel 5	1	0	0	0	0	1	0	0	0	0	1	0	0	0	0	1	0	0	0	0	1	0
22 tone Orwell-9	9	Orwel 9	1	0	0	1	0	1	0	0	1	0	1	0	0	1	0	1	0	0	1	0	1	0
22 tone Orwell-13	13	Orwel13	1	1	0	1	0	1	1	0	1	0	1	1	0	1	0	1	1	0	1	0	1	0
22 tone SuperPythagorean	12	SuPytha	1	1	0	0	1	1	1	0	0	1	1	0	0	1	1	1	0	0	1	1	0	0
22 tone Miller's Porcupine-7 Major	7	MillMPo	1	0	0	1	0	0	0	1	0	0	1	0	0	1	0	0	1	0	0	1	0	0
22 tone Jones's Porcupine-8	8	JonesPo	1	0	0	1	0	0	1	1	0	0	1	0	0	1	0	0	1	0	0	1	0	0
Alternate Proper Decatonic	10	AltProD	1	0	0	1	1	0	0	1	0	1	0	1	0	1	0	0	1	0	1	0	1	0
Exotic Symmetrical Decatonic	10	ExotSym	1	0	1	0	1	0	0	1	1	0	0	1	0	1	0	1	0	0	1	1	0	0
Standard Pentachordal Major	10	StaPenM	1	0	1	0	1	0	0	1	0	1	0	1	0	1	0	0	1	0	1	0	1	0
Static Symmetrical Major	10	StcSymM	1	0	1	0	1	0	0	1	0	1	0	1	0	1	0	1	0	0	1	0	1	0
Alternate Pentachordal Major	10	AltPenM	1	0	1	0	0	1	0	1	0	1	0	1	0	1	0	1	0	0	1	0	1	0
Dynamic Symmetrical Major	10	DynSynM	1	0	1	0	0	1	0	1	0	1	0	1	0	1	0	0	1	0	1	0	1	0
Static Symmetrical Minor	10	StaSymm	1	0	1	0	1	0	1	0	0	1	0	1	0	1	0	1	0	1	0	0	1	0
Alternate Pentachordal Minor	10	AltPenm	1	0	1	0	1	0	1	0	1	0	0	1	0	1	0	1	0	1	0	0	1	0
Dynamic Symmetrical Minor	10	DynSymm	1	0	1	0	1	0	1	0	1	0	0	1	0	1	0	1	0	1	0	1	0	0
Major quasi-equal Heptatonic	7	quHeptM	1	0	0	0	1	0	0	1	0	0	1	0	0	1	0	0	1	0	0	1	0	0
Minor quasi-equal Heptatonic, Miller's Porcupine-7	7	quHeptm	1	0	0	1	0	0	1	0	0	1	0	0	0	1	0	0	1	0	0	1	0	0
Harmonic Whole-Tone	6	HarmWho	1	0	0	0	1	0	0	1	0	0	1	0	0	0	1	0	0	0	1	0	0	0
Nine-Limit Consonant WholeTone	6	NineLim	1	0	0	0	1	0	0	1	0	0	0	1	0	0	0	1	0	0	1	0	0	0
22 tone Blues	6	22Blues	1	0	0	0	0	0	1	0	0	1	1	0	0	1	0	0	0	0	0	1	0	0
22 tone mode of Tamil Matra	7	TamilMa	1	0	0	0	1	0	0	0	1	0	0	1	0	1	0	0	0	1	0	0	1	0
Raga Kanakangi	7	raKanaK	1	0	1	0	0	1	0	0	0	1	0	0	0	1	0	1	0	0	1	0	0	0
Raga Ramkali	8	raRamka	1	1	0	0	0	0	0	1	0	1	1	0	0	1	1	0	0	0	0	0	1	0
Raga Kharaharapriya, Bhimpalasi	7	raKhara	1	0	0	0	1	0	1	0	0	1	0	0	0	1	0	0	0	1	0	1	0	0
Twenty-two tone Natural Minor, Darbari Kanada, Gandhaara Grama (Damodara), Raga Darbari, Darbari Kanada	7	raDarba	1	0	0	0	1	0	1	0	0	1	0	0	0	1	0	1	0	0	0	1	0	0
Raga Vibhas (marva)	5	raVibha	1	1	0	0	0	0	0	1	0	0	0	0	0	1	0	0	1	0	0	0	0	0
Raga Saveri	5	raSaver	1	0	1	0	0	0	0	0	0	1	0	0	0	1	1	0	0	0	0	0	0	0
Raga Deskar	5	raDeska	1	0	0	1	0	0	0	1	0	0	0	0	0	1	0	0	1	0	0	0	0	0

Raga Suddha Malhar	5	raSuddh	1	0	0	1	0	0	0	0	0	1	0	0	0	1	0	0	1	0	0	0	0	0
Raga Bhupali	5	raBhupa	1	0	0	0	1	0	0	1	0	0	0	0	0	1	0	0	0	1	0	0	0	0
Raga Hamsadhvani	5	raHamsa	1	0	0	0	1	0	0	1	0	0	0	0	0	1	0	0	0	0	0	0	1	0
Raga Durga	5	raDurga	1	0	0	0	1	0	0	0	0	1	0	0	0	1	0	0	0	1	0	0	0	0
Raga Malkauns	5	raMalka	1	0	0	0	0	0	1	0	0	1	0	0	0	0	0	1	0	0	1	0	0	0
Raga Gurjari Todi	6	raGurja	1	1	0	0	0	1	0	0	0	0	0	1	0	0	1	0	0	0	0	0	0	1
Raga Bauli	6	raBauli	1	0	1	0	0	0	0	1	0	0	0	0	0	1	0	1	0	0	0	0	1	0
Raga Kambhoji	6	raKambh	1	0	0	0	1	0	0	1	0	1	0	0	0	1	0	0	1	0	0	0	0	0
Raga Takka	6	raTakka	1	0	0	0	0	0	1	0	0	1	0	0	0	1	0	1	0	0	0	0	1	0
Raga Tilang	6	raTilan	1	0	0	0	0	0	0	1	0	1	0	0	0	1	0	0	0	0	1	0	1	0
Raga Bilashkhani Todi	7	raBilas	1	1	0	0	0	1	0	0	0	1	0	0	0	1	1	0	0	0	1	0	0	0
Raga Asavari	7	raAsava	1	1	0	0	0	1	0	0	0	1	0	0	0	1	1	0	0	0	0	1	0	0
Raga Rampurmat Pilu	7	raRampu	1	1	0	0	0	1	0	0	0	1	0	0	0	1	1	0	0	0	0	0	0	1
Raga Varali	7	raVaral	1	1	0	0	0	1	0	0	0	0	0	0	1	1	1	0	0	0	0	0	1	0
Todi That	7	todiTha	1	1	0	0	0	1	0	0	0	0	0	0	1	1	1	0	0	0	0	0	0	1
Raga Lalita	7	raLalit	1	1	0	0	0	0	0	1	0	1	0	1	0	0	1	0	0	0	0	0	0	1
Bhairav That	7	bahiTha	1	1	0	0	0	0	0	1	0	1	0	0	0	1	1	0	0	0	0	0	0	1
Gandhaara Grama (Somanatha)	7	GandSom	1	0	1	0	0	0	1	0	0	1	0	0	1	0	0	1	0	0	0	1	0	0
Murchhana Harinasva, Bhairavi That	7	Murchha	1	0	1	0	0	0	1	0	0	1	0	0	0	1	0	1	0	0	0	1	0	0
Gandhaara Grama (Popley), Murchhana Matsarikrita	7	GandPop	1	0	1	0	0	0	1	0	0	0	1	0	0	1	0	1	0	0	0	1	0	0
Raga Multani	7	raMulta	1	0	1	0	0	0	1	0	0	0	0	0	1	1	1	0	0	0	0	0	0	1
Raga Marva	7	raMarva	1	0	1	0	0	0	0	1	0	0	0	1	0	1	0	0	0	1	0	0	1	0
Purvi That, Raga Shri	7	purvTha	1	0	1	0	0	0	0	1	0	0	0	0	1	1	0	1	0	0	0	0	1	0
Raga Puriya Kalyan	7	raPuriy	1	0	1	0	0	0	0	1	0	0	0	0	1	1	0	0	1	0	0	0	1	0
Gandhaara Grama (Sarngadeva der. ma-grama)	7	GandMag	1	0	0	1	0	1	0	0	0	1	0	0	1	0	1	0	0	0	1	0	0	0
Gandhaara Grama (Sarngadeva der. sa-grama)	7	GandSag	1	0	0	1	0	1	0	0	0	1	0	0	1	0	0	1	0	0	1	0	0	0
Murchhana Hrishyaka	7	MurchHr	1	0	0	1	0	0	0	1	0	1	0	0	0	1	0	0	1	0	1	0	0	0
Raga Jogiya	7	raJogiy	1	0	0	1	0	0	0	1	0	1	0	0	0	1	0	0	1	0	0	0	1	0
22 tone Minor, Superpyth-7, Raga Jaunpuri	7	raJaunp	1	0	0	0	1	1	0	0	0	1	0	0	0	1	1	0	0	0	1	0	0	0
22 tone "Just" Minor, Asavari That	7	asavTha	1	0	0	0	1	0	1	0	0	1	0	0	0	1	0	1	0	0	1	0	0	0
Gandhaara Grama (Narada der. ma-grama), Murchhana Pauravi	7	GandNaM	1	0	0	0	1	0	1	0	0	0	1	0	0	1	0	1	0	0	0	1	0	0
Gandhaara Grama (Narada der. sa-grama)	7	GandNaS	1	0	0	0	1	0	1	0	0	0	1	0	0	1	0	0	1	0	0	1	0	0
Murchhana Suddhasadja, Raga Harikambhoji, Palaiyazh	7	MurchSu	1	0	0	0	1	0	0	1	0	1	0	0	0	1	0	0	1	0	1	0	0	0
22 tone "Just" Major, Madhyama Grama, Bilaval That, Murchhana Sud-dhamadhya	7	Madhyam	1	0	0	0	1	0	0	1	0	1	0	0	0	1	0	0	1	0	0	0	1	0
Khamaj That	7	khamTha	1	0	0	0	1	0	0	1	0	1	0	0	0	1	0	0	0	1	1	0	0	0
Shadja Grama, Murchhana Uttaramandra, Shuddha Swara Saptaka	7	Shadiag	1	0	0	0	1	0	0	1	0	1	0	0	0	1	0	0	0	1	0	0	1	0
Murchhana Sauviri, Kalyan That	7	MurchSa	1	0	0	0	1	0	0	1	0	0	0	1	0	1	0	0	0	1	0	0	1	0
Murchhana Asvakranta	7	MurchAs	1	0	0	0	1	0	0	0	1	0	0	1	0	1	0	0	0	1	0	0	1	0
Raga Yaman	7	raYaman	1	0	0	0	1	0	0	0	1	0	0	0	1	1	0	0	0	1	0	0	0	1
Raga Saurashtra	8	raSaura	1	1	0	0	0	0	0	1	0	1	0	0	0	1	1	0	1	0	0	0	1	0
Raga Bhatiyar	8	raBhati	1	0	1	0	0	0	0	1	0	1	0	0	1	1	0	0	1	0	0	0	1	0
Raga Mukhari	8	raMukha	1	0	0	1	0	1	0	0	0	1	0	0	0	1	0	1	1	0	1	0	0	0
Raga Anandabhairavi	8	raAnand	1	0	0	0	1	0	1	0	0	1	0	0	0	1	0	1	0	1	0	1	0	0
Raga Mian Ki Malhar	8	raMianK	1	0	0	0	1	0	1	0	0	1	0	0	0	1	0	0	0	1	0	1	0	1
Raga Suddha Kalyan	8	raSuddh	1	0	0	0	1	0	0	1	0	1	0	1	0	1	0	0	1	0	0	0	1	0
Raga Yaman Kalyan	8	raYamKa	1	0	0	0	1	0	0	1	0	1	0	0	1	1	0	0	1	0	0	0	0	1
Raga Gaud Sarang	8	raGaudS	1	0	0	0	1	0	0	1	0	1	0	0	1	1	0	0	0	1	0	0	1	0
Raga Bihag	8	raBihag	1	0	0	0	1	0	0	1	0	1	0	0	1	1	0	0	0	1	0	0	0	1
Raga Khamaj	8	raKhama	1	0	0	0	1	0	0	1	0	1	0	0	0	1	0	0	1	0	1	0	0	1
Raga Ramdasi Malhar	9	raRamMa	1	0	0	0	1	0	1	0	1	1	0	0	0	1	0	0	0	1	0	1	0	1
Raga Chayanat	9	raChaya	1	0	0	0	1	0	0	1	0	1	0	0	1	1	0	0	1	0	1	0	0	1
Modern Indian gamut	12	MoIndGa	1	0	1	0	1	0	1	1	0	1	0	1	0	1	0	1	0	1	0	1	1	0
Old Indian gamut	12	OlIndGa	1	0	0	1	0	1	1	1	0	1	0	0	1	1	0	0	1	0	1	1	1	0
Murchhana Abhirudgata, Kafi That, Raga Bageshri	7	MurchAb	1	0	0	1	0	1	0	0	0	1	0	0	0	1	0	0	1	0	1	0	0	0
Raga Madhuvanti, Ambika	7	raMadhu	1	0	0	0	1	0	1	0	0	0	0	1	0	1	0	0	0	1	0	0	1	0
	22	User 1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
	22	User 2	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
	22	User 3	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
	22	User 4	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1

Table 23: 24-EDO and other 24-tone-per-octave temperaments.

		tone	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23
		cents	0	50	100	150	200	250	300	350	400	450	500	550	600	650	700	750	800	850	900	950	1000	1050	1100	1150
description	t/oct	display																								
Chromatic	24	Chroma	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
Enharmonic Mixolydian	7	en Mixo	1	1	1	0	0	0	0	0	0	0	1	1	1	0	0	0	0	0	0	0	1	0	0	0
Enharmonic Lydian	7	en Lydi	1	1	0	0	0	0	0	0	0	1	1	1	0	0	0	0	0	0	0	1	0	0	0	1
Enharmonic Phrygian	7	en Phry	1	0	0	0	0	0	0	0	1	1	1	0	0	0	0	0	0	0	1	0	0	0	1	1
Enharmonic Dorian	7	en Dori	1	1	1	0	0	0	0	0	0	0	1	0	0	0	1	1	1	0	0	0	0	0	0	0
Enharmonic Hypolydian	7	enhLydi	1	1	0	0	0	0	0	0	0	1	0	0	0	1	1	1	0	0	0	0	0	0	0	1
Enharmonic Hypophrygian	7	enhPhry	1	0	0	0	0	0	0	0	1	0	0	0	1	1	1	0	0	0	0	0	0	0	1	1
Enharmonic Hypodorian	7	enhDori	1	0	0	0	1	1	1	0	0	0	0	0	0	0	1	1	1	0	0	0	0	0	0	0
Soft Diatonic Mixolydian	7	sd Mixo	1	0	1	0	0	1	0	0	0	0	1	0	1	0	0	1	0	0	0	0	1	0	0	0
Soft Diatonic Lydian	7	sd Lydi	1	0	0	1	0	0	0	0	1	0	1	0	0	1	0	0	0	0	1	0	0	0	1	0
Soft Diatonic Phrygian	7	sd Phry	1	0	0	0	0	1	0	1	0	0	1	0	0	0	0	1	0	0	0	1	0	1	0	0
Soft Diatonic Dorian	7	sd Dori	1	0	1	0	0	1	0	0	0	0	1	0	0	0	1	0	1	0	0	1	0	0	0	0
Soft Diatonic Hypolydian	7	sdhLidi	1	0	0	1	0	0	0	0	1	0	0	0	1	0	1	0	0	1	0	0	0	0	1	0
Soft Diatonic Hypophrygian	7	sdhPhry	1	0	0	0	0	1	0	0	0	1	0	1	0	0	1	0	0	0	0	1	0	1	0	0
Soft Diatonic Hypodorian	7	sdhDori	1	0	0	0	1	0	1	0	0	1	0	0	0	0	1	0	1	0	0	1	0	0	0	0
Neutral Diatonic Mixolydian, Maqam Ouchairan-Hussaini, Bayatan	7	nd Mixo	1	0	0	1	0	0	1	0	0	0	1	0	0	1	0	0	1	0	0	0	1	0	0	0
Neutral Diatonic Lydian, Dastgah-e Sehgah	7	nd Lydi	1	0	0	1	0	0	0	1	0	0	1	0	0	1	0	0	0	1	0	0	0	1	0	0
Neutral Diatonic Phrygian, Arabic Diatonic, Maqam Rast, Quasi-equal Heptatonic	7	nd Phry	1	0	0	0	1	0	0	1	0	0	1	0	0	0	1	0	0	0	1	0	0	1	0	0
Neutral Diatonic Dorian, Maqam Hussaini, Ushaq	7	nd Dori	1	0	0	1	0	0	1	0	0	0	1	0	0	0	1	0	0	1	0	0	1	0	0	0
Neutral Diatonic Hypolydian, Maqam Sikah (Segah)	7	ndhLidy	1	0	0	1	0	0	0	1	0	0	0	1	0	0	1	0	0	1	0	0	0	1	0	0
Neutral Diatonic Hypophrygian	7	ndhPhry	1	0	0	0	1	0	0	0	1	0	0	1	0	0	1	0	0	0	1	0	0	1	0	0
Neutral Diatonic Hypodorian, Miha'il Musaqa's mode: Egypt, Dastgah-e Sehgah, Maqam Nairuz	7	ndhDori	1	0	0	0	1	0	0	1	0	0	1	0	0	0	1	0	0	1	0	0	1	0	0	0
Diatonic + Enharmonic Diesis Mixolydian	7	dd Mixo	1	1	0	0	0	0	1	0	0	0	1	1	0	0	0	0	1	0	0	0	1	0	0	0
Diatonic + Enharmonic Diesis Lydian	7	dd Lydi	1	0	0	0	0	1	0	0	0	1	1	0	0	0	0	1	0	0	0	1	0	0	0	1
Diatonic + Enharmonic Diesis Phrygian	7	dd Phry	1	0	0	0	1	1	0	0	0	0	1	0	0	0	1	0	0	0	1	1	0	0	0	0
Diatonic + Enharmonic Diesis Dorian	7	dd Dori	1	1	0	0	0	0	1	0	0	0	1	0	0	0	1	1	0	0	0	0	1	0	0	0
Diatonic + Enharmonic Diesis Hypolydian	7	ddhLidy	1	0	0	0	0	1	0	0	0	1	0	0	0	1	1	0	0	0	0	1	0	0	0	1
Diatonic + Enharmonic Diesis Hypophrygian	7	ddhPhry	1	0	0	0	1	0	0	0	1	1	0	0	0	0	1	0	0	0	1	1	0	0	0	0
Diatonic + Enharmonic Diesis Hypodorian	7	ddhDori	1	0	0	0	1	1	0	0	0	0	1	0	0	0	1	1	0	0	0	0	1	0	0	0
Chromatic/Enharmonic Mixolydian	7	ce Mixo	1	1	0	0	1	0	0	0	0	0	1	1	0	0	1	0	0	0	0	0	1	0	0	0
Chromatic/Enharmonic Lydian	7	ce Lydi	1	0	0	1	0	0	0	0	0	1	1	0	0	1	0	0	0	0	0	1	0	0	0	1
Chromatic/Enharmonic Phrygian	7	ce Phry	1	0	0	0	0	0	1	1	0	0	1	0	0	0	0	0	1	0	0	0	1	1	0	0
Chromatic/Enharmonic Dorian	7	ce Dori	1	1	0	0	1	0	0	0	0	0	1	0	0	0	1	1	0	0	1	0	0	0	0	0
Chromatic/Enharmonic Hypolydian	7	cehLydi	1	0	0	1	0	0	0	0	0	1	0	0	0	1	1	0	0	1	0	0	0	0	0	1
Chromatic/Enharmonic Hypophrygian	7	cehPhry	1	0	0	0	0	0	1	0	0	0	1	1	0	0	1	0	0	0	0	0	1	1	0	0
Chromatic/Enharmonic Hypodorian	7	cehDori	1	0	0	0	1	1	0	0	1	0	0	0	0	0	1	1	0	0	1	0	0	0	0	0
Neutral Mixolydian, Iced Blizzard	7	ne Mixo	1	0	0	1	0	0	0	1	0	0	1	0	0	1	0	0	0	1	0	0	1	0	0	0
Neutral Lydian, Iced Major	7	ne Lydi	1	0	0	0	1	0	0	1	0	0	1	0	0	0	1	0	0	1	0	0	0	1	0	0
Neutral Phrygian, Iced Locrian	7	ne Phry	1	0	0	1	0	0	1	0	0	0	1	0	0	1	0	0	0	1	0	0	1	0	0	0
Neutral Dorian, Iced Fridgian, Misaelides 2nd Byzantine mode, Maqam Sikah Baladi, Maqamic-7	7	ne Dori	1	0	0	1	0	0	0	1	0	0	1	0	0	0	1	0	0	1	0	0	0	1	0	0
Neutral Hypolydian, Iced Lydian, Mohajira-7	7	nehLydi	1	0	0	0	1	0	0	1	0	0	0	1	0	0	1	0	0	0	1	0	0	1	0	0
Neutral Hypophrygian, Iced Mixolydian	7	nehPhry	1	0	0	1	0	0	0	1	0	0	1	0	0	0	1	0	0	1	0	0	1	0	0	0
Neutral Hypodorian, Iced Dark Lydian	7	nehDori	1	0	0	0	1	0	0	1	0	0	0	1	0	0	1	0	0	1	0	0	0	1	0	0
Ratio 1:2 Hemiolic Chromatic Mixolydian	7	rh Mixo	1	1	0	1	0	0	0	0	0	0	1	1	0	1	0	0	0	0	0	0	1	0	0	0
Ratio 1:2 Hemiolic Chromatic Lydian	7	rh Lydi	1	0	1	0	0	0	0	0	0	1	1	0	1	0	0	0	0	0	0	1	0	0	0	1
Ratio 1:2 Hemiolic Chromatic Phrygian	7	rh Phry	1	0	0	0	0	0	0	1	1	0	1	0	0	0	0	0	0	1	0	0	0	1	1	0
Ratio 1:2 Hemiolic Chromatic Dorian	7	rh Dori	1	1	0	1	0	0	0	0	0	0	1	0	0	0	1	1	0	1	0	0	0	0	0	0
Ratio 1:2 Hemiolic Chromatic Hypolydian	7	rhhLydi	1	0	1	0	0	0	0	0	0	1	0	0	0	1	1	0	1	0	0	0	0	0	0	1
Ratio 1:2 Hemiolic Chromatic Hypophrygian	7	rhhPhry	1	0	0	0	0	0	0	1	0	0	0	1	1	0	1	0	0	0	0	0	0	1	1	0
Ratio 1:2 Hemiolic Chromatic Hypodorian	7	rhhDori	1	0	0	0	1	1	0	1	0	0	0	0	0	0	1	1	0	1	0	0	0	0	0	0
Anchihoye: Ethiopia	5	Ethiopi	1	0	1	0	0	0	0	0	0	0	1	0	0	1	0	0	0	0	0	1	0	0	0	0
Spondeion	5	Spondei	1	0	0	1	0	0	0	0	0	0	1	0	0	0	1	0	0	1	0	0	0	0	0	0
Godzilla-5	5	Godzill	1	0	0	0	1	0	0	0	0	1	0	0	0	0	1	0	0	0	0	1	0	0	0	0
Quasi-equal Pentatonic, Semaphore-5	5	quasiPe	1	0	0	0	0	1	0	0	0	0	1	0	0	0	1	0	0	0	0	1	0	0	0	0
de Vries 5-tone	5	deVrie5	1	0	0	0	0	0	0	0	0	1	0	1	0	1	0	0	0	0	0	0	0	0	1	0
Spondeiakos	6	Spondko	1	1	1	0	0	0	0	0	0	0	1	0	0	0	1	0	1	0	0	0	0	0	0	0
Maqam Nawa	7	mq Nawa	1	0	1	0	0	0	1	0	0	0	1	0	0	0	1	0	0	1	0	0	1	0	0	0
Second plagal Byzantine Liturgical mode	7	Liturgi	1	0	1	0	0	0	0	0	0	1	1	0	0	0	1	0	1	0	0	0	0	0	0	1
Maqam Higaz-kar	7	mqHigaz	1	0	1	0	0	0	0	1	0	0	1	0	0	0	1	0	1	0	0	0	0	1	0	0
Maqam 'Ushshaq Turki, Urfa, Isfahan, Dastgah-e Shur	7	mqUshsh	1	0	0	1	0	0	1	0	0	0	1	0	0	0	1	0	1	0	0	0	1	0	0	0
Maqam Nahfat	7	mqNahfa	1	0	0	1	0	0	1	0	0	0	1	0	0	0	1	0	0	0	1	0	1	0	0	0
Maqam Saba	7	mq Saba	1	0	0	1	0	0	1	0	1	0	0	0	0	0	1	0	1	0	0	0	1	0	0	0
Maqam Sabr Jadid	7	mqSabrJ	1	0	0	1	0	0	1	0	1	0	0	0	0	0	1	0	1	0	0	0	0	0	1	0
Maqam Qarjighar, Bayati Shuri	7	mqQarji	1	0	0	1	0	0	1	0	0	0	1	0	1	0	0	0	0	0	1	0	1	0	0	0
Maqam Hizam (Huzzam, El Houzam), Rahat al Arouah	7	mqHizam	1	0	0	1	0	0	0	1	0	1	0	0	0	0	0	1	0	1	0	0	0	1	0	0
Maqam Su'ar, Naghmeh Abuata, Naghmeh Afshari	7	mqSu'ar	1	0	0	1	0	0	0	1	0	0	0	1	0	1	0	0	0	1	0	0	0	1	0	0
Dastgah-e Homayun	7	DastHom	1	0	0	1	0	0	0	0	1	0	1	0	0	0	1	0	1	0	0	0	1	0	0	0
Dastgah-e Chahargah, Athanasopoulos' Byzantine Liturgical Chromatic,	7	DastCha	1	0	0	1	0	0	0	0	1	0	1	0	0	0	1	0	0	1	0	0	0	0	1	0
Maqam 'Awg 'ara (Aug-ara)	7	mqAwgAr	1	0	0	1	0	0	0	0	0	1	1	0	0	0	0	1	0	1	0	0	0	0	0	1
Maqam Buselik	7	mqBusel	1	0	0	0	1	1	0	0	0	0	1	0	0	0	1	0	1	0	0	0	0	0	1	0
Dastgah-e Nava, Maqam Ushaq Masri	7	DastNav	1	0	0	0	1	0	1	0	0	0	1	0	0	0	1	0	0	1	0	0	0	0	1	0
Naghmeh Esfahan	7	Naghmeh	1	0	0	0	1	0	1	0	0	0	1	0	0	0	1	0	0	1	0	0	0	0	1	0
Maqam Neuter	7	mqNeute	1	0	0	0	1	0	1	0	0	0	0	0	1	0	1	0	1	0	0	0	0	1	0	0
Maqam Suznak (Soznak)	7	mqSuzna	1	0	0	0	1	0	0	1	0	0	1	0	0	0	1	0	1	0	0	0	0	0	1	0
Dance scale of Yi people: China	7	YiPeopl	1	0	0	0	1	0	0	1	0	0	1	0	0	0	1	0	0	0	1	0	1	0	0	0
Maqam Mahur	7	mqMahur	1	0	0	0	1	0	0	1	0	0	1	0	0	0	1	0	0	0	1	0	0	0	1	0

Daniel-mode of Spanish-Arab Jews	7	DanielM	1	0	0	0	1	0	0	0	1	0	1	0	0	1	1	0	0	0	1	0	0	0	0	0
Maqam Jahargah (Jiharkah), Naghmeh Bayat-e Tork, Naghmeh Dashti	7	maJahar	1	0	0	0	1	0	0	0	1	0	1	0	0	0	1	0	0	0	1	0	0	1	0	0
Maqam 'Ajam Murassah	7	maAjamM	1	0	0	0	1	0	0	0	1	0	0	1	0	0	1	0	0	0	1	0	0	0	1	0
Maqam Bayati	8	maBayat	1	0	0	1	0	0	1	0	0	0	1	0	0	0	1	0	1	1	0	0	1	0	0	0
Maqam Saba	8	ma8Saba	1	0	0	1	0	0	1	0	1	0	0	0	0	0	1	0	1	0	0	0	1	0	1	0
Maqam Mansuri	8	maMansu	1	0	0	1	0	0	1	0	1	0	1	0	0	0	1	0	0	1	0	0	1	0	0	0
Maqam Rast, Dilkashidah, Dilnishin	8	ma Rast	1	0	0	0	1	0	0	1	0	0	1	0	0	0	1	0	0	0	1	0	1	1	0	0
Maqam Suzidil 'ara	8	maSuzid	1	0	0	0	1	0	0	1	1	0	1	0	0	0	1	0	0	0	1	0	1	0	0	0
Maqam Rahat al-Arwah	8	maRahat	1	0	0	1	0	0	0	1	0	1	0	0	0	0	0	1	0	1	0	0	0	1	0	1
Maqam Iraq	8	ma Iraq	1	0	0	1	0	0	0	1	0	0	1	0	0	1	0	0	0	1	0	0	0	1	0	1
Maqam Hijaz	8	maHijaz	1	0	1	0	0	0	0	0	1	0	1	0	0	0	1	0	1	1	0	0	1	0	0	0
Maqam Musta'ar	8	maMusta	1	0	0	1	0	0	0	1	0	0	0	1	0	1	1	0	0	1	0	0	0	1	0	0
Maqam Farahnak	8	maFarah	1	0	0	1	0	0	0	1	0	0	0	1	0	0	0	1	0	1	0	0	0	1	0	1
Maqam Bastanikar, Tarz Nuin	8	maBasta	1	0	0	1	0	0	0	1	0	0	1	0	0	1	0	1	0	0	0	0	0	1	0	1
Maqam Farah Faza, Maqam Nakriz	8	maFarFa	1	0	0	0	1	0	1	0	0	0	0	0	1	0	1	0	0	0	1	0	1	1	0	0
Maqam Jabburi	8	maJabbu	1	0	0	1	1	0	1	0	0	0	1	0	0	0	1	0	1	0	0	0	1	0	0	0
Dalmonte 8-tone	8	Dalmont	1	1	0	0	0	1	0	0	0	1	0	1	0	0	0	1	0	0	0	1	0	0	0	1
Spongework	8	Spongew	1	1	0	0	0	0	1	0	0	1	1	0	0	0	0	1	0	0	1	1	0	0	0	0
Freivald Lament	8	Freival	1	1	0	0	1	0	0	1	0	0	0	1	0	0	1	0	0	0	1	0	0	1	0	0
Progressive Enneatonic	9	ProgEnn	1	1	0	1	0	0	1	0	0	0	1	0	0	0	1	1	0	1	0	0	1	0	0	0
Triforce-9	9	Triforc	1	0	0	1	0	0	1	0	1	0	0	1	0	0	1	0	1	0	0	0	1	0	1	0
Maqam Huzzam	9	maHuzza	1	0	0	1	0	0	0	1	0	1	0	1	0	1	0	1	0	1	0	0	0	1	0	0
de Vries 9-tone, Semaphore-9	9	deVrie9	1	0	0	0	1	1	0	0	0	1	1	0	0	0	1	1	0	0	0	1	1	0	0	0
Godzilla-9	9	Godzil9	1	0	0	0	1	0	0	0	1	1	0	0	0	1	1	0	0	0	1	1	0	0	0	1
Maqam Shawq Afza	9	maShawq	1	0	0	0	1	0	0	0	1	0	1	1	0	0	1	0	1	0	1	0	0	0	1	0
Xiangliu	9	Xiangli	1	0	0	0	1	0	0	0	1	0	1	1	0	0	1	0	0	0	1	0	0	1	1	0
Migration-10	10	Migrati	1	1	0	0	1	0	0	1	1	0	0	1	0	0	1	1	0	0	1	0	0	1	0	0
Neutral Hypolydian Decatonic, Mohajira-10	10	nhDecat	1	1	0	0	1	0	0	1	1	0	0	1	0	0	1	0	0	1	1	0	0	1	0	0
Young Decatonic, Decimal-10	10	younDec	1	0	1	0	0	1	0	1	0	0	1	0	1	0	1	0	0	1	0	1	0	0	1	0
Oljare Decatonic	10	OljaDec	1	0	1	0	0	1	0	1	0	1	0	0	1	0	1	0	0	1	0	1	0	1	0	0
Anguirus-10	10	Angui10	1	0	1	0	1	0	0	1	0	1	0	0	1	0	1	0	1	0	0	1	0	1	0	0
Neutral Dorian Decatonic, Maqamic-10	10	ndDecat	1	0	0	1	1	0	0	1	0	0	1	1	0	0	1	0	0	1	0	0	1	1	0	0
Breed Decatonic	10	BreedDe	1	0	0	1	1	0	0	1	0	0	1	0	0	1	1	0	0	1	0	0	1	1	0	0
Maqam Yakah	10	maYakah	1	0	0	0	1	0	0	1	1	0	1	1	0	0	1	0	0	0	1	0	1	1	0	0
Maqam Basandida	10	maBasan	1	0	0	0	1	0	1	1	0	0	1	0	1	0	1	0	0	0	1	0	1	1	0	0
Maqam Shawq Tarab	10	maShawq	1	0	1	1	0	0	1	0	1	0	1	0	0	0	1	0	1	0	0	0	1	0	1	0
Freivald-11	11	Freiv11	1	0	0	1	0	1	0	1	0	1	0	1	0	0	1	0	1	0	1	0	1	0	1	0
Maqam Hayyan	11	maHayya	1	0	0	0	1	0	1	1	0	0	1	0	1	0	1	0	1	0	1	0	0	1	1	0
Hemiaug-12	12	Hemiaug	1	0	0	0	0	1	1	1	1	0	0	0	0	1	1	1	1	0	0	0	0	1	1	1
Iceface	12	Iceface	1	0	0	1	1	0	0	1	1	0	1	0	0	1	1	0	0	1	1	0	0	1	1	0
Vaisvil's Mixed-Quarters	12	Vaisvil	1	1	0	0	1	0	1	1	0	0	1	1	0	0	1	0	1	1	0	1	0	0	1	0
Freivald-13	13	Freiv13	1	1	0	1	0	1	0	1	0	1	0	1	1	0	1	0	1	0	1	0	1	0	1	0
de Vries 13-tone	13	deVri13	1	1	0	1	0	1	0	1	0	1	0	1	0	1	1	0	1	0	1	0	1	0	1	0
Agmon Diatonic DS5	13	AgmonDi	1	0	1	0	1	0	1	0	1	0	1	1	0	1	0	1	0	1	0	1	0	1	0	1
Young Half-Octave Diatonic, Decimal-14	14	young14	1	0	1	0	1	1	0	1	0	1	0	1	1	0	1	0	1	1	0	1	0	1	0	1
Anguirus-14	14	Angui14	1	0	1	1	0	1	0	1	1	0	1	0	1	0	1	1	0	1	0	1	1	0	1	0
Godzilla-14	14	Godzi14	1	0	0	1	1	0	0	1	1	1	0	0	1	1	1	0	0	1	1	1	0	0	1	1
Triforce-15	15	Trifo15	1	1	0	1	1	0	1	0	1	1	0	1	1	0	1	0	1	1	0	1	1	0	1	0
	24	User 1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
	24	User 2	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
	24	User 3	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
	24	User 4	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1

13 CHANGE LOG

This section contains the list of the firmware updates that required additions or modifications to the manual.

V. 152

Extended the Store Pattern functionality to the *Length* parameter. Now USTA stores all the channels of the pattern, not only the one selected.

Changed the shortcut for the channel Hold function: hold SET ALL and the desired channel – *CV A*, *CV B*, *Gate A* or *Gate B*. (The previous combination was: hold ESC and the desired channel.)

Added external PPM (Pulses per Minute) value to the Dashboard to display how many pulses are received in a minute by the external clock input (up to 3000).

Added the capability for the Dashboard to display the currently playing stage in Performance Mode. To activate it, enter the Project Menu, scroll until the ShowInPlay option, and select Yes (by default, it is set to No). When engaged, USTA will use the fourth row of the Dashboard (which in Edit Pattern mode displays the last edited stage) to display in real-time all the five stage values. This option might slow down the device.

Rearranged the Dashboard design to include the total pattern length (in units).

Added the Mute Channel function: hold ESC and press te desired channel (*CV A*, *CV B*, *Gate A* or *Gate B*).

Added CW or CCW pattern rotation while editing it: hold the track button and rotate the navigation encoder on the selected pattern. It affects the values of *CV A*, *CV B*, *Length*, *Gate A*, *Gate B*, and the stage colors.

Added the quick initialization of all the track values:

hold the pencil button for 3 seconds in Edit Pattern to initialize to default all the stage values of the selected track:

hold the pencil button for 3 seconds in Edit Song to initializes to default all the song slots of the selected track.

Added infinite stage loop:

set the stage loop length to 0 (now accessible):

disable stage loop and continue the pattern by double-clicking SET ALL.

Added cross-clone layer between *CV A* & *CV B* and *Gate A* & *Gate B*: hold the target layer button and press the corresponding encoder to apply the clone to a specific pattern

Added a new auxiliary gate target: Run. It uses the incoming gate high/gate low signals to start, stop, and reset USTA with external devices. Four possible configurations are possible, named Run 1, Run 2, Run 3, Run 4. To activate it, enter the Project Menu, scroll until the Aux Target option, and select any of the four modes.

V. 150

Added fast change of first/last pattern settings in Performance Mode on the selected track: hold *Set All* (B.1) and push an encoder for the first pattern and another for the last pattern.

Added the cloning of first/last pattern settings, in performance mode, from the selected track to the others, by double-clicking *Shift All* (B.2).

Navigating through stages in composition mode updates the stage values on the display across *CV A*, *CV B*, length, *Gate A* and *Gate B*.

14 TECHNICAL DATA

14.1 SPECIFICATIONS

Parameter	Details		Min	Typ	Max	Unit
Current Draw		+12V			260	mA
		-12V			50	
Size				36		HP
Input Details	CV	Impedance		> 90		KΩ
		Resolution		12		bit
		Amplitude steps		~4.15		mV
		Voltage span	-5		10	V
	Gate	Impedance		>100		KΩ
		Minimum Trigger Amplitude		> 1.5		V
		Minimum Pulse Period	800			µs
Output Details	CV	Impedance		< 50		Ω
		Resolution		16		bit
		Amplitude steps		~0.26		mV
		Voltage span	-5		10	V
	Gate	Impedance		< 50		Ω
		Amplitude		10		V

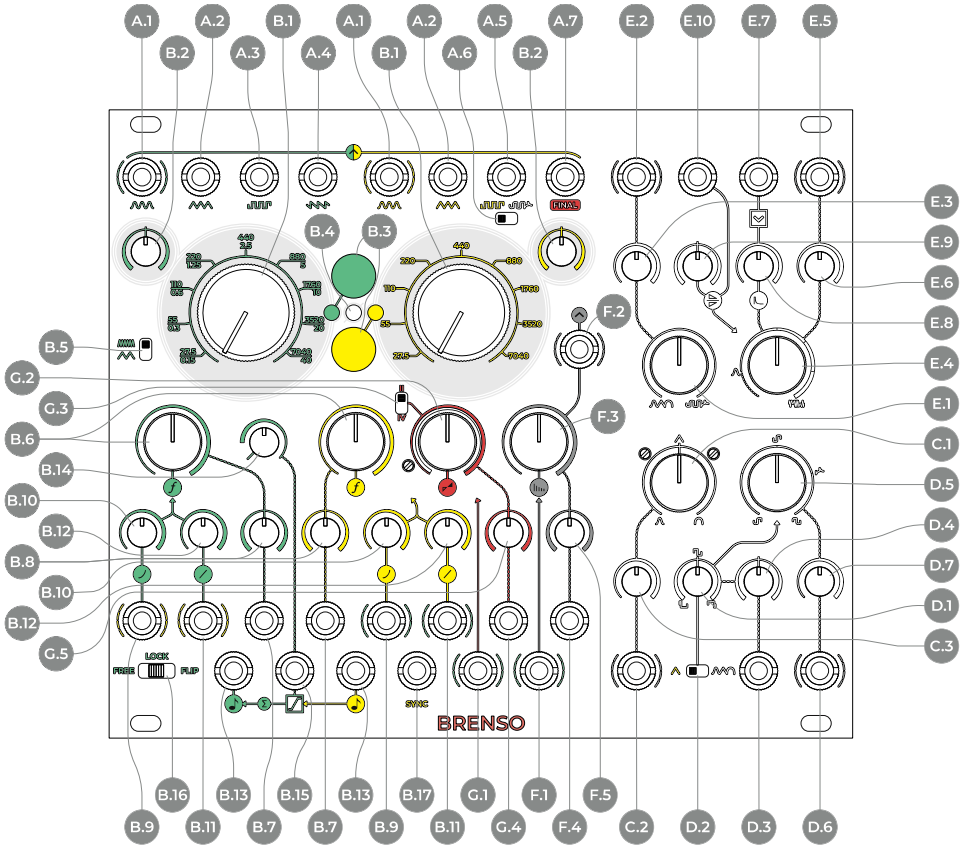


Figure 87: BRENSO interface.

A Outputs
A.1 Sine Wave Output
A.2 Triangle Wave Output
A.3 Square Wave Output
A.4 Sawtooth Wave Output
A.5 Square/Shaped Pulse Output
A.6 Square/Shaped Pulse Switch
A.7 Final Output
B Frequencies
B.1 Coarse Frequency
B.2 Fine Frequency
B.3 Coarse Frequency Lock
B.4 Phase Relationship LED
B.5 Frequency Scale Switch
B.6 Frequency Modulation (FM) Deviation
B.7 FM Deviation CV Input
B.8 FM Deviation CV Attenuverter
B.9 Exponential FM Input
B.10 Exponential FM Attenuator
B.11 Linear Through-Zero FM Input
B.12 Linear TZFM Attenuator
B.13 V/Oct Input
B.14 V/oct Integrator Level
B.15 V/oct Integrator CV Input
B.16 Sync Switch
B.17 Sync Input
C Triangle Shaper
C.1 Triangle Shaper
C.2 Triangle Shaper CV Input
C.3 Triangle Shaper CV Attenuator
D Square Shaper
D.1 Pulse Width
D.2 Pulse-Width Modulation (PWM) Source
D.3 PWM CV Input
D.4 PWM CV Attenuverter
D.5 Pulse Shaper
D.6 Pulse Shaper CV Input
D.7 Pulse Shaper CV Attenuator
E Wavefolder
E.1 Wavefolder Source
E.2 Wavefolder Source CV Input
E.3 Wavefolder Source Attenuator
E.4 Wavefolder
E.5 Wavefolder CV Input
E.6 Wavefolder Attenuator
E.7 Wavefolder Ping Input
E.8 Wavefolder Ping Decay
E.9 Wavefolder Symmetry
E.10 Wavefolder Symmetry CV Input
F Modulation Bus
F.1 Modulation Bus Input
F.2 Modulation Bus Output
F.3 Modulation Bus Level
F.4 Modulation Bus Level CV Input
F.5 Modulation Bus Level CV Attenuverter
G Amplitude
G.1 Amplitude Modulation/Ring Modulation (AM/RM) Input
G.2 AM/RM
G.3 AM/RM Switch
G.4 AM/RM CV Input
G.5 AM/RM CV Attenuverter

1 PHILOSOPHY, DESIGN AND SIGNAL FLOW

BRENSO is Frap Tools' primary analog source of articulated audio waveforms whose degree of entanglement can be precisely set by the musician.

Its concept developed from a reflection on the very meaning of the word 'complex', often used to describe this kind of oscillators after the famous Buchla 259 definition. Complex comes from the Latin verb plector, literally meaning 'to braid', or 'to weave.' The purpose of BRENSO is to update the usual approach to 'complex oscillators' by offering many threads to be woven together, rather than a pre-defined plait of controls and waveforms: this to improve clarity, manageability, and to offer more sonic options to the artist.

We designed a unique signal flow from scratch, in order to expand the modulation routing and offer to the musician more access to the crucial parts of the circuit.

BRENSO's architecture consists of three main parts: two oscillators, which can modulate each other's frequency, a timbre modulation section, dedicated to waveshaping, PWM and wavefolding, and a final stage of amplitude modulation. These sections will be described in the next chapters of this manual: Frequency, Timbre, and Amplitude.

2 FREQUENCY

BRENSO generates sounds with two oscillators, whose pitch can be independently regulated. Their frequencies can modulate each other (linearly and exponentially – the linear FM is Thru-Zero), or they can be synced (with Flip Sync or Lock).

2.1 OSCILLATORS

The two sound sources are analog, triangle-core oscillators with excellent stability and tracking. The oscillators are labeled with green and yellow graphics on the front panel, respectively.

Each oscillator features four outputs for different wave shapes. The output jacks are located at the top of the front panel. The green oscillator's outputs are, from left to right: sine (A.1), triangle (A.2), square (A.3), and sawtooth (A.4); the yellow one's are sine (A.1), triangle (A.2), square/shaped square (A.5), plus a *Final* jack socket (A.7) that outputs the sine and/or the Shaped Square after their processing through the *Timbre* and *AM* sections.

2.1.1 Fine and Coarse Tuning

The two largest knobs of the front panel (B.1) are the main control for the two oscillators' *Coarse* frequency. Their range goes from ~27.5Hz to ~7040Hz, as displayed by the graphics.

The *Fine Frequency* knob at the top left of the green oscillator and the top right of the yellow one (B.2) finely adjusts the frequency by adding or subtracting >1 semitones.

The green oscillator can work at sub-audio rates as well, through the *Frequency Scale* switch (B.5). The graphics around the main knob display the LFO frequency range below the audio-rate one: when set in 'Low' scale, the green oscillator goes from ~0.15Hz to ~40Hz, ~176 times slower.

Please note that the oscillators are trimmed to match as close as possible the usage at audio rate: a consequence of this is that the green oscillator may not perfectly match the graphics when used as LFO due to components tolerance (see below §6.2).

A red LED (B.4) visually displays the acoustic beats generated by the two oscillators' frequencies.

2.1.2 Coarse Frequency Lock

Sometimes the frequency knobs can be endlessly tweaked in a creative way during the performance, but some other times, they just need to do what they were originally designed for: tuning an oscillator before the performance, and nothing more. However, especially in crowded patches, the Frequency knob of an oscillator can be hit by mistake, causing an unwanted detune and jeopardizing the whole performance – especially when more than one oscillator is playing polyphonic melodies!

To avoid this unfortunate circumstance, we designed a digital implementation that can be engaged to prevent accidental twists of the knobs from detuning the oscillators: the *Coarse Frequency Lock*.

It consists of two buttons (B.3), one per each oscillator, that, when pushed, digitally sample and hold the voltage of the *Coarse Frequency* knob (B.1), thus keeping the oscillator's frequency steady. The two buttons are equipped with an LED: when the circuit is engaged, they will light up green or yellow, respectively. From this moment on, the *Coarse Frequency* knob will no longer change the frequency when moved, and the oscillator's pitch will be changed only through external CV or internal frequency modulation. To restore the knob's function, push the button again: if the Coarse knob has changed position in the meantime, the oscillator's coarse frequency will set accordingly.

Please note that the *Fine Frequency* knob (B.2) is not affected by the Frequency Lock, and it will always be available in case a little adjustment is needed (for example, to compensate small detuning in extreme climatic situations).

Please note that the Frequency Lock offers a way to avoid unintended touches: you still need to warm up your BRENSO as for any other module before to use it.

2.1.3 V/oct and Integrator

The frequency of each oscillator can be externally controlled through dedicated *V/oct* inputs (B.13): an external voltage patched to these inputs will offset the frequency set by the *Coarse* and *Fine Frequency* knobs (B.1 and B.2).

Excellent tracking is guaranteed over more than 6 octaves: this feature provides not only good intonation for particularly spread melodic lines but also a more precise ratio between the two oscillators, which translates into richer and better harmonics when any modulation between them is engaged, especially frequency modulation, which will be discussed in the next chapter.

The two oscillators can be independently controlled with different voltages. However, it is also possible to apply the same CV to both oscillators by patching it to the yellow *V/oct* input and routing it to the green oscillator through the *V/oct Integrator* (B.14), without further patching.

This circuit applies the same CV patched into the yellow CV input to the green oscillator after a linear integration, whose amount is set by the knob position (B.14).

When the *Integrator* knob is at its leftmost position, the green oscillator is not affected by the yellow CV, because it would take an infinite time to reach the target voltage. When the integrator knob is at its rightmost position, the exact same voltage offset of the yellow oscillator is applied to the green one, with a very fast integration. When the knob is set to any other position, a time lag, similar to a glide effect, is applied to the voltage routed to the green oscillator: on the left, the glide effect will be longer, and it will be increasingly short as the knob is rotated clockwise. Please remember that the CV is always the same: the only difference is the time required for the green oscillator to reach the value.

It is possible to combine the voltage routed through the *V/oct Integrator* with another offset applied to the green *V/oct* input: for example, the two oscillators can be used in unison and controlled with the same CV through the integrator, while an *n+1* voltage coming from SAPÈL is patched to the green *V/oct Input* to randomly shift the green oscillator one or more octaves higher.

Furthermore, it is also possible to modulate the integration time with external CV through the *V/oct Integrator CV Input* (B.15): in this case, any external voltage will be summed to or subtracted from the current position of the knob.

2.2 Frequency Modulation

The two oscillators of the BRENSO can be frequency-modulated, even at an audio rate. Such modulation can be linear or exponential (or both at the same time). You can use external sources to modulate the oscillators' frequency, but every oscillator's input is semi-normalled to the other oscillator's sine wave.

When an oscillator's frequency is modulated at a sub-audio rate, this generates noticeable fluctuations of the pitch, similar to a vibrato effect.

When the modulating signal runs at audio rate, the human ear can no longer perceive the fluctuations: instead, the result of audio-rate *frequency modulation* (FM) is a more complex sound whose timbre is a result of the interaction of the two frequencies (the one of the oscillator being modulated, which is usually called 'carrier', and the one of the 'modulator').

The change in timbre is due to the generation of other frequencies, called 'sidebands,' which are the sums and the difference of the carrier and the integer multiples of the modulator. If the ratio of the carrier frequency and the modulating frequency is an integer number, such as 3:1, the sidebands generated by FM will be harmonic, i.e., they will be integer multiples of the carrier and modulating frequencies. If the ratio is expressed by a non-integer number, the sidebands will be inharmonic, i.e., non-integer multiples of the carrier and modulating frequencies. This latter circumstance produces the bell-like sounds often associated with this technique.

FM in the analog domain is often an approximate process, because of the difficulty for the analog components to guarantee a precise ratio between carrier and modulator frequencies.

The number and amplitude of sidebands is proportional to the amount of modulation that is applied to the carrier, which is often called 'deviation': this value defines the difference between the carrier's frequency and the higher or lower frequency that it reaches when modulated. The more deviation, the wider will be the fluctuations of the carrier frequency, and the greater the number of sidebands.

The relation between the deviation and the modulator's frequency, both expressed in Hz, defines the FM Index. (For example, if the modulator's frequency is 200Hz and the deviation is 400Hz, the FM index would be 400/200=2.)

BRENSO allows you to control the FM deviation, not the Index: the reason is that the deviation is expressed in Hz, so its impact over the carrier's frequency will become exponentially lower as the latter increases. This generates sounds that are rich in harmonics in the low and mid range, without becoming excessively harsh in the highest range (See §2.2.1).

Depending on how the modulation is applied to the carrier signal, FM can be *exponential* or *linear*. Linear FM modulates the carrier on the basis of the frequency: in other words, in linear FM, the modulator increases and decreases the carrier frequency by the same Hz value, according to the modulation amount. Exponential FM modulates the carrier on the basis of its frequency, i.e., with intervals: a symmetric bipolar signal will thus increase and decrease the carrier frequency by the same

interval (for example one octave), according to the modulation amount.

The main difference between these two techniques is that Linear FM generates sidebands which are equally spaced above and below the carrier frequency, while exponential FM does not. This happens because the exponential modulation is asymmetrical: if an A=440Hz waveform is modulated exponentially, and the modulation amount is +/-1 octave, the carrier frequency will oscillate between 220Hz and 880Hz, which is 220Hz below and 440Hz above the original frequency. Such modulation also causes a shift of the central frequency: in this case, it will be 550Hz, which is exactly 330Hz above 220Hz and below 880Hz. This, in turn, will generate a perceived detune of the original pitch, which will be different every time the carrier frequency is changed.

Sidebands are the sum and difference of the carrier and integer multiples of the modulator. There can be some cases, however, in which the difference between the carrier and the modulator would provide a negative number. Since negative frequencies are not physically possible, these sidebands are usually inaudible. For example, if the carrier frequency is 150Hz and the modulator frequency is 200Hz, the first couple of sidebands would be at 350Hz and −50Hz. However, a conventional analog oscillator stops oscillating whenever it reaches 0Hz, thus removing part of the spectra.

For this purpose, an approach called Thru-Zero FM has been implemented: with this technique, the negative sidebands (the ones located "below zero") are generated, but with an inverted phase. The result is a richer and more natural musical timbre with even less pitch drifting than may occur with analog FM.

2.2.1 FM Routing

Both oscillators of the BRENSO can work as carrier and modulator at the same time: this means that the green oscillator can modulate the yellow one, which in turn modulates back the green. This technique allows creating extremely complex sounds with just two oscillators, whose final spectral content may even reach the realm of noise.

To achieve this result, BRENSO is equipped with two FM buses, one for the yellow and one for the green oscillator (B.7 to B.12). The scope of each bus is to offer an explicit modulation routing with advanced capabilities.

Each FM bus has three main controls: the big one is the FM *Deviation* knob (B.6) connected to its *CV Input* (B.7) and *Attenuverter* (B.8), while the other two smaller knobs are the Linear TZFM Attenuator, and the Exponential FM attenuator. The Deviation knob sets the overall modulation that it is applied to the oscillator, while the two *attenuators* (B.10, B.12) determine the specific amount of *Linear Through-Zero (TZ)* and *Exponential FM*. All the knobs increase the value through a clockwise rotation, starting from the leftmost position, where no modulating signal is routed.

By default, the linear and exponential modulation source of each oscillator is the other one: the green oscillator is thus semi-normalled to the yellow *Linear TZFM* and *Exponential FM* inputs (B.9 and B.11), and vice versa. It is also possible to break the normalization and use another signal by patching it to the desired input. In this case, the respective attenuator knob will attenuate the external signal amplitude. It is thus possible to use up to four different modulation sources at the same time.

In order to produce audible effects, one must set both the *Deviation* knob and either the *Linear TZ* or *Exponential attenuator* to a value higher than zero (the leftmost position).

This bus design offers two great advantages: on the one hand, it allows combining linear and exponential FM independently per each oscillator. On the other, by having independent CV inputs on the two buses, it is possible to control the modulation amount over each oscillator with different sources, thus creating more articulated timbres.

2.3 Sync

Syncing refers to a variety of techniques originally developed to improve and stabilize the relative intonation of two or more analog oscillators.

The common ground is that one oscillator should be used as a reference to which other oscillators must be compared and, if different, corrected: different correction techniques translate into different syncing circuits.

It soon became clear, however, that in certain sync circuits, an exaggerated modulation of a "slave" oscillator introduced pleasant overtones to the final sound, and these techniques became widely employed in sound synthesis to generate more complex timbres. This is the case with, for example, Hard Sync, which is often implemented in sawtooth-core oscillators. This circuit takes two oscillators, called 'master' and 'slave', and forces the slave's waveform to reset back to 0 at each master's duty cycle. Modulating the slave oscillator's frequency will result in a rich sound which changes timbre without changing pitch since the waveform still resets at the master's rate. The downside of this process is that every time the slave wave resets and thus drops back to the beginning of the duty cycle, it may produce a sort of unpleasant "spike," which becomes more and more noticeable as the modulation gets deeper.

BRENSO is equipped with two different circuits, which provide different results and are designed to accomplish different tasks: they are called *Lock* and *Flip Sync*. They can be selected through a three-way switch for the green oscillator (B.16), and via a jumper on the back of the PCB for the yellow oscillator.

2.3.1 Lock

The *Lock* circuit is designed to provide a precise but subtle correction of an oscillator's pitch (the 'slave') when it is very close to an integer multiple or divisor of another oscillator's frequency (the 'master'). It is mainly used to compensate for slight tracking variations that may occur when CV controlling more oscillators with the same V/oct signal.

The *Lock* system uses the master's square wave to slightly change the thresholds of the slave oscillator's core: it will shift them up when the master's waveform is positive, and down when it is negative. As a result, the slave oscillator will gently rush and drag to follow the master's frequency, without resets or abrupt changes in the waveform direction.

Since this circuit is designed to correct very small differences in frequency, we recommend using it mainly when the slave oscillator is within a semitone from its desired pitch. If the ratio between the two oscillators is not an integer number, some changes in the harmonic spectra may occur.

By default, the green oscillator can be locked to the yellow one by moving the three-way *Sync Switch* (B.16)to the *Lock* position.

The yellow oscillator does not have any hardwired slave capability, but it can be synced to an external waveform by patching it to the *Sync Input* (B.17) and setting the back jumper to *Lock*.

2.3.2 Flip Sync

It has been said that sometimes an extreme modulation of the synced oscillator may be deliberately used to create complex waveforms, as with Hard Sync. BRENSO's triangle cores allow a different technique, typical of this kind of oscillator, called *Flip Sync* (or *Reverse Sync*): instead of forcing the slave oscillator's waveform back to the beginning at every master's duty cycle, it reverses the wave direction. This translates into a more mellow tone, which allows creative modulation without hearing the harsh "spikes" made by the slave oscillator's waveform reset by sawtooth-based Hard Sync.

Flip Sync is activated by moving the *Sync Switch* of the green oscillator's section (B.16) to its rightmost position: when engaged, the green core will invert its waveform direction at every yellow duty cycle.

As we said above, *Flip Sync* can be activated for the yellow oscillator, too, by placing the jumper on the back of the PCB to the Sync position. In this way, the yellow oscillator will become the slave of any signal patched to its *Sync Input* (B.17).

This technique, in contrast to *Lock*, creates dramatic changes in the slave's waveform and can be used for more expressive and creative purposes. Since it is not phase-based, and since preserving the original waveform is not a priority, there is no preferred frequency range for setting the slave oscillator to be synced properly. Of course, a master frequency higher than the slave one would result in a change of amplitude, of the slave since it will not always be possible to reach the full cycle without a change of direction.

3 TIMBRE

So far, we have seen several ways of generating complex timbres by just acting on the oscillators' frequencies, but BRENSO is capable of much more. A whole section is dedicated to modulating the waveforms of the yellow oscillator through a series of circuits. Here is an overview of the signal routing.

· The yellow triangle wave is routed to a waveshaper called *Triangle Shaper* (C.1, C.2, C.3), which morphs between a sine and a logarithmic wave. The resulting waveform is sent to two destinations: a crossfader called *Source* (E.1, E.2, E.3), and a comparator for *Pulse-Width Modulation* (PWM – D.1, D.3, D.4).

· The comparator takes the modulated waveform and derives a pulse-wave, which can be modulated; alternatively, the *Pulse-Width Modulation* (PWM) *Source* can be set to be the pure triangle wave of the yellow oscillator (D.2).

· The PWM wave is sent to another waveshaper called *Pulse Shaper* (D.5, D.6, D.7), which emphasizes the high or low harmonics. The resulting signal is sent to two destinations, the *Source* crossfader, and the square wave output of the yellow oscillator. A *switch* (A.6) below the jack socket (A.5) selects the output source: pure square wave, or shaped-wave.

· The *Sources* crossfader (E.1, E.2, E.3) receives signals from the *Triangle Shaper* and the *Square Shaper* and blends them together. The result is fed into a *Wavefolder* (E.4 to E.10).

· The *Wavefolder* further processes the sound, and its output is sent to another crossfader for amplitude modulation (see below, §4) before being routed to the Final output (A.7).

All the aforementioned waveshaping techniques are performed over the yellow oscillator's waveforms, and their amount can be modulated (even at audio rate) by the green oscillator's sine wave, through a circuit called *Modulation Bus* (F.1 to F.5). The following paragraphs will describe the modulation circuits, and the last one will describe the role of the green oscillator and the *Modulation Bus*.

3.1 Triangle Shaper

The first waveform modulation circuit is the *Triangle Shaper* (C.1). It can be seen as a three-way mixer that blends three waveforms: a sine wave (leftmost position), an almost-pure triangle wave (noon), and a logarithmic

waveform. (Both the sine and the log are generated by shaping the triangle wave, hence its name.)

It is possible to control this parameter, even at audio rate, by patching an external signal to the *CV input* (C.2). An *Attenuator* (C.3) allows for precise scaling of the incoming signal. By default, the modulation input is semi-normalled to the *Modulation Bus* output (F.2).

The signal coming from this circuit is then routed to the *Source* crossfader (E.1) and then possibly to the *Wavefolder* (E.4), which can be heard through the final output (more on this below, §3.3).

3.2 Pulse Shaper

The other waveshaping circuit consists of two sections strictly connected: first, a comparator generating a pulse wave and capable of pulse-width modulation (PWM), and then another waveshaper to which the PWM output is routed.

3.2.1 Pulse-Width Modulation (PWM)

The PWM circuit generates a pulse-wave through a comparator. This technique requires a waveform and a voltage value used as a reference. The comparator, as the name suggests, compares the waveform to the reference voltage: every time the waveform is equal to or higher than the reference voltage, the output voltage will be high; conversely, every time the waveform is lower than the reference voltage, the output will be low. The alternation of high and low voltages produces a pulse waveform, whose width depends, on the one hand, on the initial waveform fed into the comparator, and on the other, on the value of the reference voltage.

A classic example of pulse-width modulation is achieved by using a fixed triangle waveform and varying the comparator threshold: this operation changes the reference voltage and thus the point where the output voltages are high and low, but without affecting the duty cycle, i.e. the frequency.

It has been said that both the elements of the comparator (the waveform and the reference voltage) play a fundamental role in a pulse-wave generation: the two main controls of BRENSO's PWM circuit affect exactly these two parameters.

The two-way *Source* switch at the bottom (D.2) selects the source to be fed into the comparator: when it is set on the left, the source is yellow triangle wave, straight from the oscillator's core, and the result is a classic pulse wave; when it is set on the right, the source is the wave generated by the *Triangle Shaper* according to its knob position (it can be a sine, a logarithmic wave or everything in between).

The *Pulse Width* knob (D.1) sets the comparator threshold and thus varies the pulse wave, or the ratio between the positive and negative side of the waveform (also called 'symmetry'). When the source is the triangle wave, and the knob is centered, the ratio will be of ~50%, which translates into an almost perfect square wave. Moving the knob on the left or on the right will change this ratio, thus generating a positive or negative asymmetry).

The *CV Input* at the bottom (D.3) accepts any signal to be used to vary the wave symmetry, with a dedicated *Attenuverter* (D.4) to scale or invert it: when the attenuverter is set at noon, no modulation is applied; rotate the knob on the left or right to apply a negative or positive modulation, respectively. Patch any LFO-like signal to the modulation input and adjust the attenuverter to achieve the classic PWM sound.

When the *Triangle Shaper* is used as a source for the comparator, it can produce more complex results: for instance, the pulse wave symmetry can be varied by modulating the wave shape, without changing the comparator threshold. Modulate both the waveshaper and the symmetry control to generate articulated modulations, which are harder to obtain with more conventional PWM circuits.

3.2.2 Waveshaper

The *Square Shaper* circuit is BRENSO's second waveshaper. It further shapes the harmonic content of the pulse wave generated by the PWM circuit (§ 3.2.1).

When its main knob (D.5) is at the leftmost position, it emphasizes the lower frequencies; the higher overtones will become increasingly higher until the noon position is reached: at this point, the waveform generated by the PWM circuit is almost purely reproduced. When the knob is rotated past noon, the higher frequencies will be progressively emphasized, until roughly two o'clock (75% of the knob stroke), where they will have the highest amplitude. From this point onward, the lower frequencies will be emphasized again, but with an inverted phase, until the rightmost position is reached, where the signal is the same as at the leftmost position, but with inverted phase.

It is possible to control this parameter, even at audio rate, by patching an external signal to the *CV input* (D.6). An *Attenuator* (D.7) allows for precise scaling of the incoming signal. By default, the modulation input is semi-normalled to the *Modulation Bus* output (F.2).

The output of the *Pulse Shaper* is routed to two different points of the circuit. First, to the yellow square wave output (A.5), where it can be selected with a dedicated switch (A.6), instead of the regular, core-derived square wave; then, it is sent to the final wavefolding stage.

3.3 Wavefolder

So far, we have described two signal paths: the one of the *Triangle* Shaper, and the one of the PWM through the

Square Shaper. Both are further modulated through a *Wavefolder*.

3.3.1 Sources

The first stage of the wavefolding circuit is the *Source* control. It is essentially a crossfader between the two waveshaped signals: the one from the *Triangle* Shaper and the other from the PWM through the *Square Shaper*.

When its knob (E.1) is set at the leftmost position, only the signal coming from the *Triangle Shaper* is sent to the *Wavefolder*. Rotate the knob clockwise to blend in the signal coming from the Square Shaper: at noon, the blend will be 50-50. At the rightmost position, however, the signal sent to the waveshaper will not be purely the one coming from the Square Shaper, but a small amount of the other one will still be audible.

This design choice has been made to improve the tonal characteristics of the folded signal. Because of its nature, any "pure" pulse waveform may produce some irrelevant results wen folded, or even some amplitude loss. To prevent this from happening, the Source control will always retain some of the triangle-shaped signal, which will dramatically improve the behavior of the wavefolder.

If you want to hear the pure sound of the Pulse Shaper, you can always use the *Square/Shaped Pulse Output* (A.5) and set its *Source* switch (A.6) to the leftmost position.

It is possible to control this parameter, even at audio rate, by patching an external signal to the *CV input* (E.2). An *attenuator* (E.3) allows for precise scaling of the incoming signal. By default, the modulation input is semi-normalled to the *Modulation Bus* output (F.2).

3.3.2 Folding

A wavefolder is a circuit that amplifies a waveform beyond a pre-determined threshold: for this reason, it can also be classified as a distortion unit. Once its peaks reach the threshold (both on the positive and negative sides), instead of being clipped, they are "folded" on themselves. When they reach the opposite threshold, they are folded back again, and the cycle repeats, generating a number of folds which depends on the circuit design. The result of these folds is an increasing number of overtones that generate a richer sound.

The main control of BRENSO's wavefolder is the *Wavefolder* knob (E.4). Its range is divided into six layout graphic portions with increasing line thickness, which roughly reflect the number of folds performed by the circuit.

The first section is marked with a dotted line and spans from the leftmost position to the icon of a "clean" waveform. In this range, the knob simply controls the amplitude of the incoming signal, from 0 to unity gain.

Rotating the knob from this point onward will generate more and more folds, until the maximum is reached, at the rightmost position.

It is possible to control this parameter, even at audio rate, by patching an external signal to the *CV input* (E.5). An *Attenuator* (E.6) allows for precise scaling of the incoming signal. By default, the modulation input is semi-normalled to the *Modulation Bus Output* (F.2).

3.3.3 Symmetry

A peculiar feature of a wavefolding circuit is that it can be forced to fold the positive half-cycle before the negative one, and vice versa, by adding a voltage bias to the incoming signal. This translates into an unbalanced distortion, which provides different overtone configurations.

This technique can be performed on BRENSO through the *Symmetry* knob (E.9): at noon, the incoming waveform is perfectly balanced. Rotate the knob clockwise to increase the folds of the positive half-cycle, and counterclockwise to increase the folds on the negative half-cycle.

A *CV Input* (E.10) sums any external voltage to the value selected by the knob.

3.3.4 Ping

In many natural sounds, a higher amplitude sound often contains a higher number of harmonics in the sound spectra. Moreover, many natural instruments have a higher amplitude when they begin to generate a note, which gradually decays over time. The consequence of these two facts is that the timbre of many acoustic musical instruments evolves while a note is played, having richer harmonic content during the attack than during the decay. In other words, the harmonic content is often vaguely proportional to the amplitude of the signal: think, for example, about guitar strings, whose tone changes according to the strength it is plucked with.

This peculiarity contributes to the dynamic range of any musical instrument, and it is often an issue with electronically generated waveforms, whose harmonic content remains the same throughout different amplitudes.

The wavefolder circuit can be very helpful for changing the harmonic content over time and achieve more dynamic results, and, for this reason, BRENSO is equipped with a circuit specifically designed to dynamically change the fold numbers over time by responding to an external impulse.

The *Ping* circuit uses an external trig patched to its input (E.7) to excite the wavefolder, then integrates it with a nonlinear decay curve. The result is that the folder, once excited, quickly opens the wavefolder circuit above its maximum value, and then gradually closes it to the level set by the *Wavefolder* knob (E.4). The amount of this decay is set through the *Ping Decay* knob (E.8): at its leftmost position, it is extremely fast, while at its rightmost position, it becomes significantly longer.

Remind that the lowest level to which the *Ping* decay ends is set by the *Wavefolder* knob: if it is set to the rightmost position, the effects of the Ping will be less noticeable.

The circuit detects any steep rising edge and uses it to excite the wavefolder. It means that trigs are preferred, but any gate signal can be used as well: the *Ping* circuit will behave in the same way, regardless of the gate length.

Theoretically, any steep transition between a low and a high voltage can be used to excite the *Ping*, but practically sometimes using signals different than trigs or gate may cause it not to work as expected.

3.4 Timbre Modulation Bus

The four parameters of this section (*Triangle Shaper*, *Pulse Shaper*, *Source*, and *Wavefolder*) can be controlled both via external CV and through an internal semi-normalized routing called *Modulation Bus*.

The *Modulation Bus* is basically a multi-target VCA circuit: its input (F.1) is semi-normalled to the green oscillator's sine wave output (A.1), and its output is semi-normalled to the four *CV Inputs* listed above (C.2, D.6, E.2, E.5). Its main knob (F.3) controls the VCA *Level*, which can also be externally controlled through a *CV input* (F.4) with a dedicated attenuverter (F.5). At its leftmost position, the main knob closes the VCA, and at its rightmost position, it reaches unity gain and outputs a sine wave almost identical to the one coming from the green oscillator's sine output.

At its most basic configuration, the *Modulation Bus* sets the amount of the green sine signal to be sent to the four Modulation Inputs: here, it can be independently regulated for each of the four sections of the circuit. For example, you can set the *Level* knob (F.3) to noon, which will route to the *CV Inputs* a sine wave of half the amplitude; then, you can set the *Pulse Shaper Attenuator* (D.7) to a subtle level, and the *Wavefolder CV Attenuator* (E.6) to perform a deeper modulation. This allows you to carefully dose the amount of internal modulation to each of the four parameters.

The main purpose of the *Modulation Bus*, however, is to dynamically control the amount of modulation sent to the four *CV Inputs* at the same time, especially via external CV. For example, you can set the *Level* knob (F.3) fully counter-clockwise to close the VCA, then patch an envelope to the *Level CV Input* (F.4) and adjust its amount through the attenuverter (F.5): in this way, the envelope will control the amount of modulation sent to the four CV inputs, which, in turn, will scale it independently to their destinations.

Further operations can be performed. First, the input semi-normalization of the *Modulation Bus* can be broken by patching any signal to the *Input* jack socket (F.1). The new signal is treated in the same way: it is routed to the four destinations, where it can be independently scaled, and it can be further modulated via the *Level CV Input* (F.4).

Then, the *Modulation Bus Output* (F.2) allows you to take any signal processed by the VCA and send it wherever you need in the patch. The most "extreme" configuration of this circuit is as a stand-alone VCA, which can process an external signal, using an external CV, and send the processed signal to a different module, without even affecting any component of the BRENSO.

Remember that the semi-normalization is broken only through the inputs: if you patch the *Modulation Bus Output* to any point of your patch, its signal will still be internally routed to the four *CV Inputs* of the *Timbre* section, until a jack is patched into them.

4 AMPLITUDE

We have described how BRENSO can generate complex timbres by modulating the frequencies or the waveforms, but there is one additional section: following the signal routing, after the waveshaping and wavefolding sections, a further degree of modulation can be engaged through the *Amplitude Modulation* section.

This section is a two- or four-quadrant linear multiplier very similar in concept and design to the one that can be found in the FALISTRI module (refer to FALISTRI's §3.2 for technical details).

One input of the multiplier is the signal coming from the *Timbre* section, while the other can be set by the musician. By default, the second input is semi-normalled to the green sine wave: patch a cable to this input to break the internal semi-normalization.

Both inputs are unattenuated 10V peak-to-peak bipolar signals: when replacing the second input, you can use any signal you like, but its overall amplitude may lead to different results.

The main control is the Amplitude Modulation/Ring Modulation (*AM/RM*) knob (G.2), which is essentially a crossfader between the signal coming from the *Timbre* section and its amplitude-modulated copy. When the knob is set fully counterclockwise, the signal from the *Final* output will be exactly the one coming from the wavefolder. Rotate it clockwise to blend in the amplitude-modulated signal, until the rightmost position is reached: here, only the signal coming from the four-quadrant multiplier will be heard.

This crossfading can be externally controlled via external CV, whose input (G.4) features a dedicated attenuverter (G.5) to scale or invert the incoming signal.

4.1 Amplitude Modulation and Ring Modulation

Just like in the FALISTRI, this multiplier can work with two or four quadrants. In simpler terms, while the signal coming from the waveshaper is always bipolar, the

modulator (semi-normalled to the green sine wave) can be either unipolar or bipolar. Further down the line, it can be said that the modulating signal can perform either Amplitude Modulation (when unipolar) or Ring Modulation (Bipolar).

The difference with FALISTRI's Four Quadrant Multiplier is that, while its modulation type changes according to the signal patched in (bipolar for RM, unipolar for AM), here the expected signal is always 10V peak-to-peak, which is then internally scaled to perform the two tasks via a dedicated switch (G.3).

At its upper position, the switch scales the modulating signal on the positive side only, thus engaging only two quadrants and performing amplitude modulation. Set it to the lower position to make it bipolar, engage the four quadrants, and perform ring modulation.

For the differences between AM and RM, please refer to its section in the FALISTRI's manual (§3.2).

5 TRIMMERS

BRENSO is equipped with 23 trimmers for calibration purposes. Some of them are meant to be operated by Frap Tools only, while others can be used by expert musicians who need to fine-tune certain parameters.

The trimmers are not manual potentiometers, therefore not designed to handle a large number of twists. For this reason, you must be exactly sure about what you want to achieve when you put your hands on a trimmer, and you should limit any trimming operation to the strict necessary.

5.1 Accessible Trimmers

There are 17 trimmers that an expert user can access, both on the front panel and on the back of the PCB. They control seven parameters, described in the following paragraphs.

5.1.1 Coarse Frequency

Underneath the two Coarse Frequency Knob, on the front panel, there are two couples of trimmers labeled *m* and *M*. They control the minimum and maximum value of the Coarse Frequency knobs of each oscillator, in order to make them match as close as possible the frequency scales of the panel label.

5.1.2 Sine Wave Symmetry

Close to the Coarse Frequency Trimmers lie two more couples of trimmers, one per each oscillator, labeled + and x: they control the symmetry and gain of the stage that shapes the triangle wave into a sine wave. By default, the sine wave is calibrated to be almost pure, but it may be interesting to alter its shape on purpose to bring in the second harmonic.

5.1.3 Sawtooth Wave Symmetry

At the upper right corner of the back of the PCB, there is a trimmer that regulates the symmetry of the green sawtooth waveform. As for the sine waves, this is factory calibrated to an almost-pure sawtooth, but it can be altered on purpose.

5.1.4 Exponential FM Zero

On the back of the PCB, two trimmers in the lower-right corner control the DC-Offset compensation for the external source of exponential frequency modulation.

5.1.5 Triangle Waveshaper Shape

Two trimmers close to the Triangle Waveshaper knob control the shape of the logarithmic waveform that can be heard when the knob is at the rightmost position. This circuit is borrowed from the FALISTRI module.

5.1.6 Wavefolder Symmetry

This trimmer is located on the back of the PCB and adjusts the symmetry of the wavefolding circuit, i.e., the number of folds that appear on the positive and negative front of the waveform at a given position of the Wavefolder knob and when the Symmetry knob is at noon.

5.1.7 Four-Quadrant Multiplier

This trimmer controls the symmetry of the amplitude and ring modulations: it is used to make sure that, whenever the modulating signal reaches 0V, the modulated one is as closest possible as its un-modulated version.

To calibrate the Amplitude Modulation circuit, patch a dummy cable to the second input of the Amplitude Modulation section, and set the modulation amount fully clockwise, and trim it to reach the lowest possible amplitude.

5.1.8 Comparator

This circuit prevents the oscillator core from stalling by controlling the direction of the negative front of the bipolar waveform. The trimmer regulates the position of the threshold below which the triangle wave is detected as negative by the oscillator core.

Due to common factors such as power or ground distribution, it may happen that an oscillator stalls when set at low frequencies and modulated with a carrier with sharp transients. In this case, gently rotate this trimmer (labelled *Core Comp*) counterclockwise with a trim pot screwdriver until the oscillator is heard again. If you have troubles with this operation contact us at support@frap.tools.

5.2 Non-Accessible Trimmers

There are six more trimmers that must not be touched for any reason, because it will void the module's warranty. They control the crucial parameters of the oscillators and are carefully calibrated by our technicians: if modified, they may cause BRENSO to behave in a wrong

way. They are all in couples (one per each oscillator core) and located on the back of the PCB.

5.2.1 Gain

This trimmer controls the gain applied to the external CV patched to the V/oct input. The gain circuit provides the right scaling of the external CV before it is routed to the expo converter.

5.2.2 Base

This trimmer regulates the offset that is added to the exponential conversion of the oscillator's control voltages. It prevents the response curve from losing linearity as it gets close to 0, and the oscillator from stalling.

5.2.3 Symmetry

This trimmer regulates the symmetry of the triangle core, from which all the other waveforms are derived.

6 TECHNICAL DATA

6.1 Simple Signal Flow

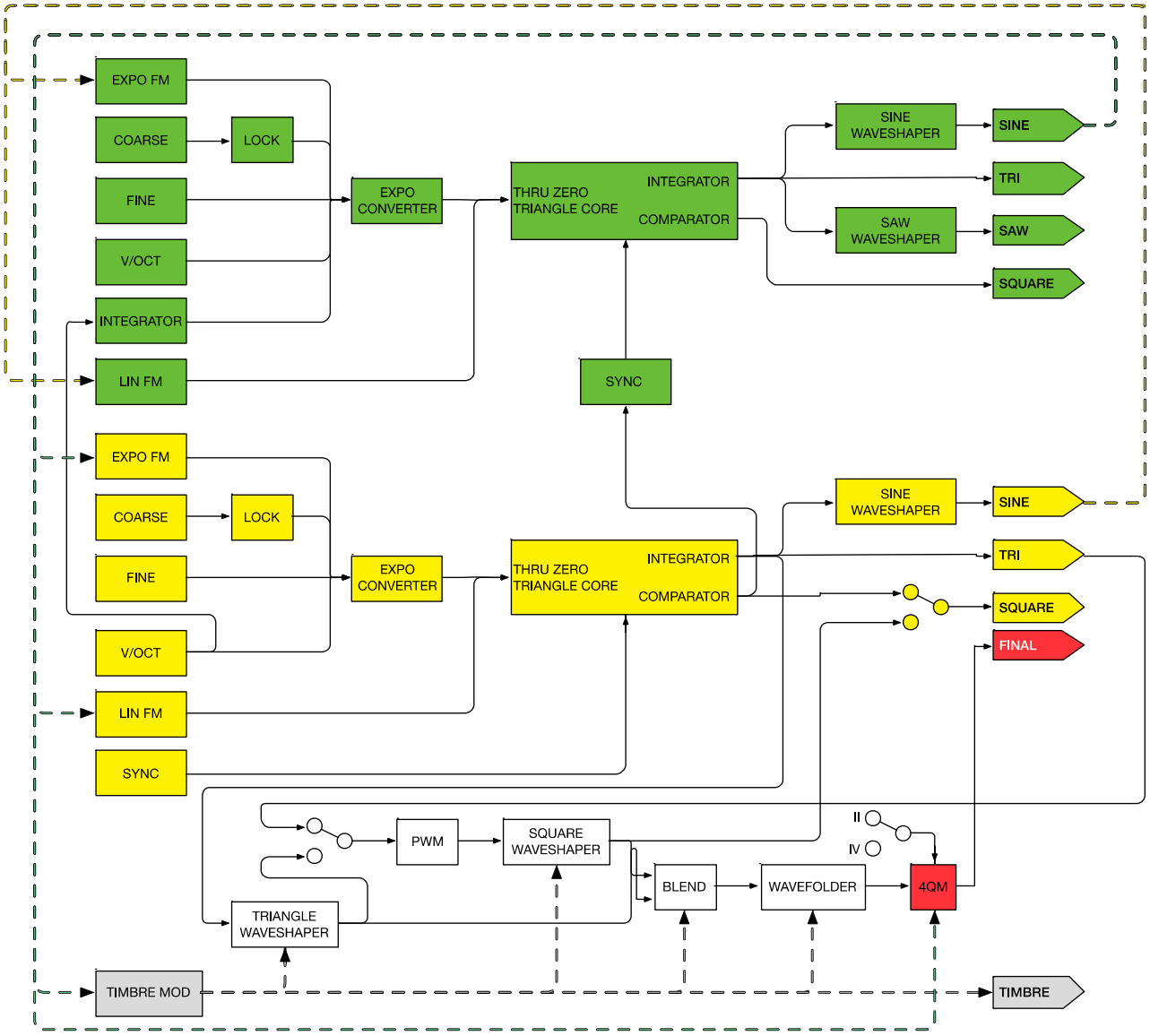


Figure 88: BRENSO's signal flow.

6.2 SPECIFICATIONS

Parameter	Details		Min	Typ	Max	Unit
Current Draw		+12V			325	mA
		-12V			235	
Size				30		HP
Input details (1)	V/oct	Impedance		>90		KΩ
	FM source			>30		
	Deviation CV			>90		
	Integrator CV			>40		
	Sync			>60		
		Minimum amplitude		2		Vpp
		Suggested amplitude		±5		
	AM CV	Impedance		>40		KΩ
	AM source			>90		
		Suggested amplitude		±5		
	Timbre Modulation source	Impedance		>40		KΩ
	Timbre modulation CV			>90		
	Triangle Waveshape CV			>60		
	Pulse Waveshaper CV			>60		
	PWM CV			>90		
	Sources CV			>25		
	Wavefolder CV			>60		
	Wavefolder Symmetry CV			>50		
	Ping CV			>90		
Output Details	Audio outputs	Impedance		>100		Ω
		Amplitude (2)		±5		Vpp
Oscillators	Knob range	Audio rate	27.5		7040	Hz
		LFO mode	0.15		10	V
			~6.6		25	ms
	Ratio (3)			176:1		
	Tracking (4)			±5		¢
	Fine tuning			>1		st
	Core crosstalk			-50		dB

(1) All buffered.
(2) Unloaded.
(3) Subject to ±3% tolerance.
(4) Measured at 440Hz over ±3 octaves.

6.3 REVISIONS

Starting from lot no. 200901 we rearranged the board layout to facilitate the calibration procedure in the lab. The functions of the module and its tech specs have not been changed.

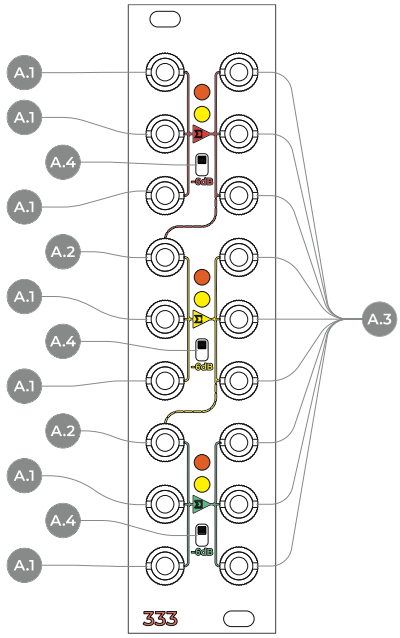


Figure 89: 333 interface.

A In/Out
A.1 Inputs
A.2 Semi-Normalled Inputs
A.3 Outputs
A.4 -6dB Switches

The 333 is an analog summing and distribution module for Eurorack modular systems. It is composed of three identical sections, each of those comprising a high-quality summing amplifier with 3 inputs, and three independently buffered outputs.

The summing amplifier circuitry and the system optimization are derived directly from the CGM creative mixer series, with the additional ability to handle signals starting from DC.

1 DESIGN

Each of the three sections, red, yellow, and green, sums up to three signals, plugged to any of its three inputs, and distribute this sum to three independently buffered outputs.

The first input of yellow and green sections (A.2) is semi-normalled to the after switch sum result of the previous section (so the yellow to the red, and the green to the yeow).

All the inputs are DC coupled, so you can use each section to sum audio signals, control voltages as well as merging clocks.

Practice the 333 in many forms with these Techniques:
Voice Spread #1
-6dB Switches
Sidechain #1

2 TECHNICAL DATA

2.1 FLOW CHART

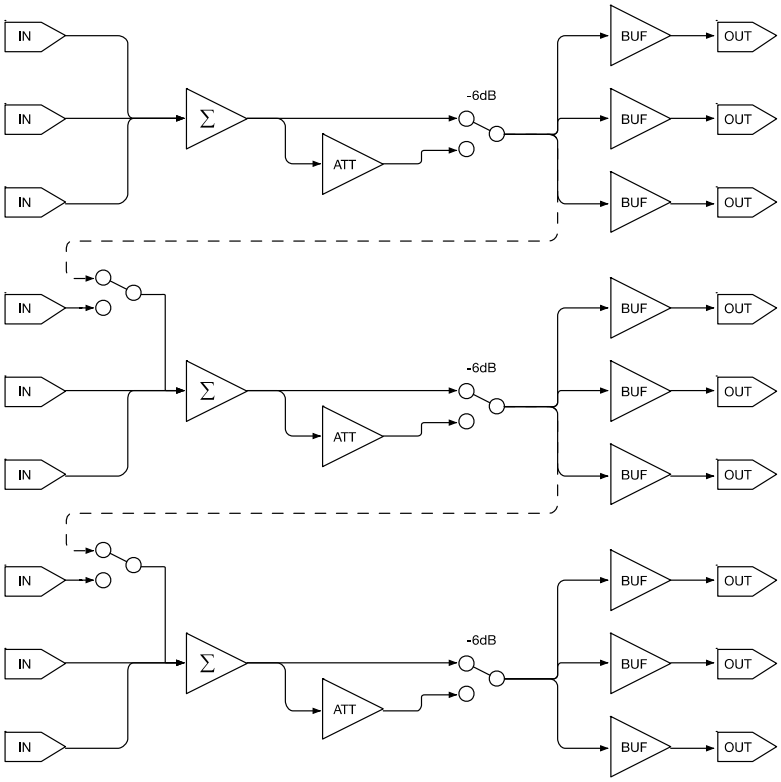


Figure 90: 333's flow chart.

2.2 SPECIFICATIONS

Parameter	Details		Min	Typ	Max	Unit
Current Draw		+12V			48	mA
		-12V			48	
Size				6		HP
Input impedance				>90		KΩ
Output impedance				<50		Ω
CV Tolerance				<0.5		%
		global offset per section	-20		20	mV
Frequency response (1)			DC		30	KHz
Harmonic Distortion (THD+N) (2)				<0.01		%

(1) Within 1dB, measured at +10dBu onto a 1KΩ load.
(2) Measured at 1KHz with output signal of +10dBu onto 1KΩ load with a 10Hz to 80KHz bandwidth measurement system.

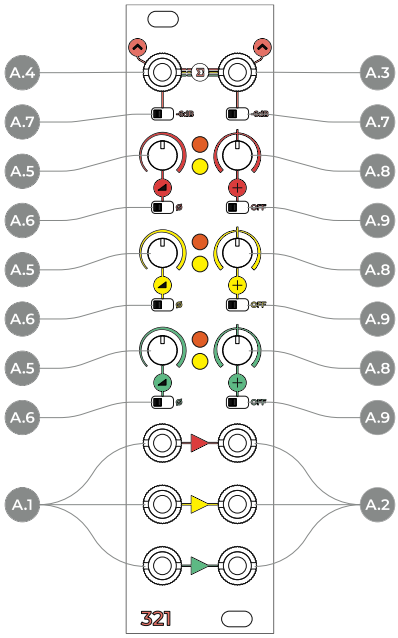


Figure 91: 321 interface.

A 321
A.1 Inputs
A.2 Outputs
A.3 Sum Output
A.4 Unpatched Sum Output
A.5 Scale
A.6 Phase Switch
A.7 -6dB Switches
A.8 Offset
A.9 Offset Switch

The 321 includes the most useful functions we always need in every patch, to edit and combine dynamically the signals in use.

It is capable of scaling any incoming signal, invert its phase, offset it, and combine with other signals. It is completely DC coupled, so may work well for both audio and CV. It can also be used as a CV source.

1 DESIGN

The 321 is composed of three identical sections, the red, yellow and green, where each of those is capable of scaling any signal, from 0 up to 2 times (+6dB), flip its phase, and apply a DC offset, allowing to shift your audio or CV up or down.

In case no signals are connected to the section input (A.1), the offset may act as a CV source too.

The three sections are then summed together into two independent summing stages: the top right jack (A.3) outputs the sum of all three sections, while the top left one (A.4) outputs the sum of the unconnected sections only (semi-normalled). The 321 is entirely DC coupled, allowing the process of control voltages as well as audio signals.

In case CVs or audio signals are particularly hot, the two 6dB attenuators (A.7), one per each summing section might be helpful to reduce exactly by 50% the incoming signals.

Practice the 321 in many forms with these Techniques:
Octave Blends
Sidechain #1
FUMANA Feedback #2
FUMANA Feedback #3

2 TECHNICAL DATA

2.1.1 Flow Chart

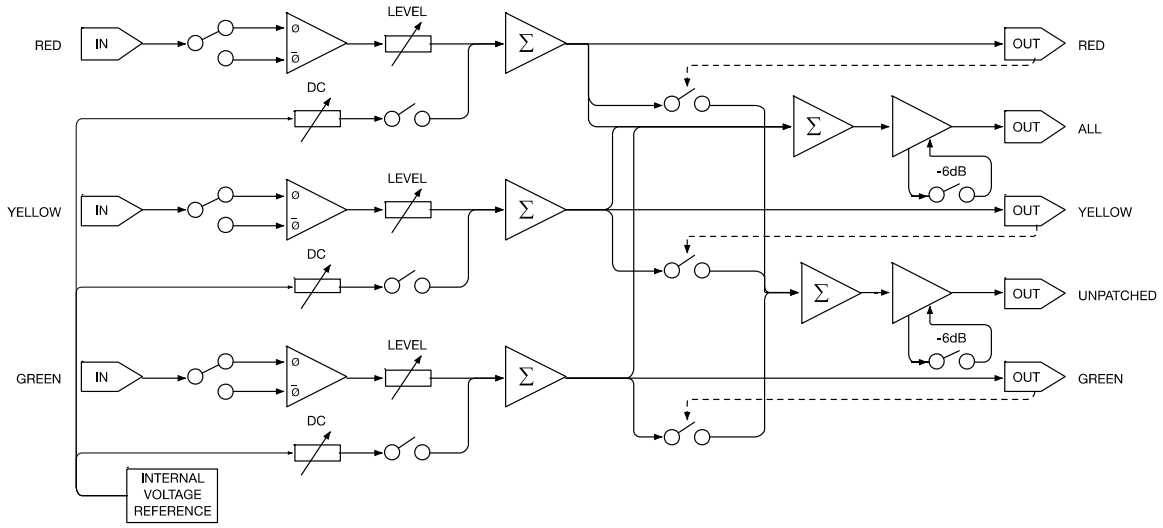


Figure 92: 321 flow chart.

2.1.2 Specifications

Parameter	Details		Min	Typ	Max	Unit
Current Draw		+12V			48	mA
		-12V			48	
Size				6		HP
Input impedance				>90		KΩ
Output impedance				<50		Ω
Frequency response (1)			DC		19	KHz

(1) Within 1dB, measured at +10dBu onto a 1KΩ load.

WHAT'S IN THE BOX

Code	Description	Q.ty	Detail
F-CAN0-MX	CGM Creative Mixer – Channel	1	Channel Module
		1	16 to 10 poles IDC Power Cable
		2	M3x6 black and zinc with plastic washer for mounting
F-GRU0-MX	CGM Creative Mixer – Group	1	Group Module
		2	M3x6 black and zinc with plastic washer for mounting
		2	10 poles IDC Link Cable – 1 Group to 4 Channels
F-MAS0-MX	CGM Creative Mixer – Master	1	Master Module
		1	16 to 10 poles IDC Power Cable
		2	M3x6 black and zinc with plastic washer for mounting
		1	10 poles IDC Link Cable – 1 Master to 1 Group
F-SAP0-RG	SAPÈL	1	SAPÈL Module
		1	16 to 10 poles IDC Power Cable
		4	M3x6 black and zinc with plastic washer for mounting
F-FUM0-SP	FUMANA	1	FUMANA Module
		1	16 to 10 poles IDC Power Cable
		4	M3x6 black and zinc with plastic washer for mounting
F-FAL0-DE	FALISTRI	1	FALISTRI Module
		1	16 to 10 poles IDC Power Cable
		4	M3x6 black and zinc with plastic washer for mounting
F-UST0-SQ	USTA	1	USTA Module
		1	16 to 10 poles IDC Power Cable
		1	microSD card
		1	microSD to SD adapter
		4	M3x6 black and zinc with plastic washer for mounting
F-BRE0-OS	BRENSO	1	BRENSO module
		1	16 to 10 poles IDC Power Cable
		4	M3x6 black and zinc with plastic washer for mounting
F-3330-SM	333	1	333 Module
		1	16 to 10 poles IDC Power Cable
		2	M3x6 black and zinc with plastic washer for mounting
F-3210-TA	321	1	321 Module
		1	16 to 10 poles IDC Power Cable
		2	M3x6 black and zinc with plastic washer for mounting
F-C120-LK	CGM Link Cable – 1 Master to 2 Groups	1	10 poles IDC Link Cable – 1 Master to 2 Groups
F-C130-LK	CGM Link Cable – 1 Master to 3 Groups	1	10 poles IDC Link Cable – 1 Master to 3 Groups
F-C140-LK	CGM Link Cable – 1 Master to 4 Groups	1	10 poles IDC Link Cable – 1 Master to 4 Groups
F-C011-LK	CGM Link Cable – 1 Group to 1 Channel	2	10 poles IDC Link Cable – 1 Group to 1 Channel [NOW DISCONTINUED]
F-C012-LK	CGM Link Cable – 1 Group to 2 Channels	2	10 poles IDC Link Cable – 1 Group to 2 Channels [NOW DISCONTINUED]
F-C013-LK	CGM Link Cable – 1 Group to 3 Channels	2	10 poles IDC Link Cable – 1 Group to 3 Channels [NOW DISCONTINUED]
F-C015-LK	CGM Link Cable – 1 Group to 5 Channels	2	10 poles IDC Link Cable – 1 Group to 5 Channels [NOW DISCONTINUED]
F-C016-LK	CGM Link Cable – 1 Group to 6 Channels	2	10 poles IDC Link Cable – 1 Group to 6 Channels [NOW DISCONTINUED]
F-C017-LK	CGM Link Cable – 1 Group to 7 Channels	2	10 poles IDC Link Cable – 1 Group to 7 Channels [NOW DISCONTINUED]
F-C018-LK	CGM Link Cable – 1 Group to 8 Channels	2	10 poles IDC Link Cable – 1 Group to 8 Channels [NOW DISCONTINUED]
F-G1C2-LS	CGM Link System – 1 Group to 2 Channel Modules	2	10 poles IDC Link Cable – 1 Group to 2 Channel Modules (C or QSC)
F-G1C4-LS	CGM Link System – 1 Group to 4 Channel Modules	2	10 poles IDC Link Cable – 1 Group to 4 Channel Modules (C or QSC)
F-G1C8-LS	CGM Link System – 1 Group to 8 Channel Modules	2	10 poles IDC Link Cable – 1 Group to 8 Channel Modules (C or QSC)

LIST OF REVISIONS

Revision	Date	Description
1	May 2019	USTA preliminary content Added Revision section SAPÈL, FALISTRI reorganized hierarchy Added Techniques section
2	Sep 2019	Updated Techniques section Improved FUMANA section Improved SAPÈL section SAPÈL, 321, 333 replaced flow chart Added Interfaces section USTA official content
3	Apr 2020	Updated Techniques section BRENSO preliminary content USTA updated to fw 152
4	Jun 2020	BRENSO official content Included in-text hyperlinks to Frap Tools' Techniques Updated layout and formatting Added indexed labelled interfaces at the beginning of every chapter to improve the readability Improved cross-references
5	Jun 2020	Fixed wrong labelling on BRENSO
6	Nov 2020	Quad Stereo Channel added to CGM section Added Revisions for Falistri and Brenso Added the section Modular Synthesis: Core Concepts Added the section Suggested Readings Improved SAPÈL's Philosophy and Design paragraph Revised the Specifications section and moved at the end of the individual module's chapters Corrected various typos Fixed wrong labelling on FALISTRI Improved cross-references