OPERATOR'S MANUAL

SAMARKANDA

RESAMPLING DELAY PERFORMANCE STATION

Model of 1953

Quadruple resampling delay · Lossless chaining of sections · Delay range of 0.5ms to 15 seconds per channel for a total of 60 seconds when stacked · Analog (tape-like) and digital (granular) sweep behaviours · Continuous, synced, and tap-tempo operation · Clock division and multiplication from 1:8 to 8:1 · Buffer freezing and 1V/oct loop playback control · Reverse delay effect · Multiple options for coupling channels

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MODULE OVERVIEW

Salut! Thank you for purchasing this Xaoc Devices product. Samarkanda [ˌsamar'kanda] is a performance-oriented digital delay station with resampling and looping capabilities. It offers four independent channels of delay with a wide range starting at 0.5ms and reaching 15 seconds per channel.

Samarkanda's channels can be losslessly stacked or chained, linked, and clocked for synchronous operation but also controlled with calibrated 1V/octave voltages. Each channel can infinitely hold and/or reverse the signal in its buffer. It can also act as a resonator, chorus or flanger. Sweeping the delay or loop length can be seamlessly switched between emulated analog (BBD/tape-like) and digital (granular-like) behaviour.

The audio path is built with high quality components, including 24-bit AKM converters and 32-bit internal processing.

Samarkanda performs its delay duties in the purest form possible, without additional filtering or other beautifying tricks.

As an open-architecture modular device, it allows you to achieve every signal modification and modulation through external patching.

INSTALLATION

The module requires 42hp worth of free space in the Eurorack cabinet. Make sure your power supply can handle the additional load (+210mA, -85mA). The ribbon-type power cable must be plugged into the bus board, paying close attention to polarity orientation. The red stripe indicates the negative 12V rail and is supposed to match the arrowhead, **–12V**, *or red stripe marks on both the unit and the bus board.*

The module itself is secured against reversed power connection; however, reversing the 16-pin header **MAY CAUSE SERIOUS DAMAGE** *to other components of your system because it will short-circuit the +12V and +5V power rails.*

To better understand the device and avoid common pitfalls, we strongly advise the user to read through the entire manual before using the module.

MODULE OVERVIEW

The front panel of Samarkanda (fig. 1) is visually divided into four identical sections: **A, B, C,** *and* **D,** *representing the four channels of the device. In each section, the big* **TIME** *rotary dial* ❶ *sets the length of the delay between 0.5 and 150 milliseconds, 5ms and 1.5 seconds, or 50ms and 15 seconds, depending on the position of the* **RANGE** *switch* ❷. *The scale of the* **TIME** *dial is exponential, offering precise adjustments of the time with resolution always proportional to the value set.*

The delay time set by the dial and the **RANGE** *switch can be modulated by external CV patched into the jack* ❸ *labeled* **TIME**. *This jack accepts voltages in the range of -5V to +10V and is scaled so that each change by 1V increases or decreases the delay time twice (which effectively means 1V/octave). Besides continuous control of the delay time, each channel may be synchronized to an external clock patched into the* **SYNC** *jack* ❹ *or manually by tapping the* **TAP TEMPO** *button* ❺.

The currently selected delay time control method is indicated by the button's backlight color (green: continuous; yellow: tap tempo; red: external sync). The period of the button's blinking is always equal to the current delay time. Note that there is another scale with discrete division and multiplication factors from 1:8 to 8:1 around the **TIME** *knob. This scale is applicable when a given channel is synced.*

The audio signal **INPUT** ❻ *is located in the upper left corner of each section, and the main* **MIX OUT** ❼ *is on the opposite side, next to the input of the subsequent channel. Note there is an arrow printed between the output of each channel and the input of the next channel, indicating that the signal is, by default, digitally passed between channels when no cable is plugged into the input, thus eliminating all artifacts resulting from unnecessary D/A and A/D conversions.*

The proportions of the input vs the delayed signal are controlled by a dedicated **MIX** *knob* ❽ *and CV patched to its associated* **MIX** *input* ❾ *that accepts bipolar voltages. Besides the* **MIX OUT**, *there is also a* **WET OUT** *jack* ❿ *for direct access to the delayed signal and for creating an external processing loop within each channel (or across channels!).*

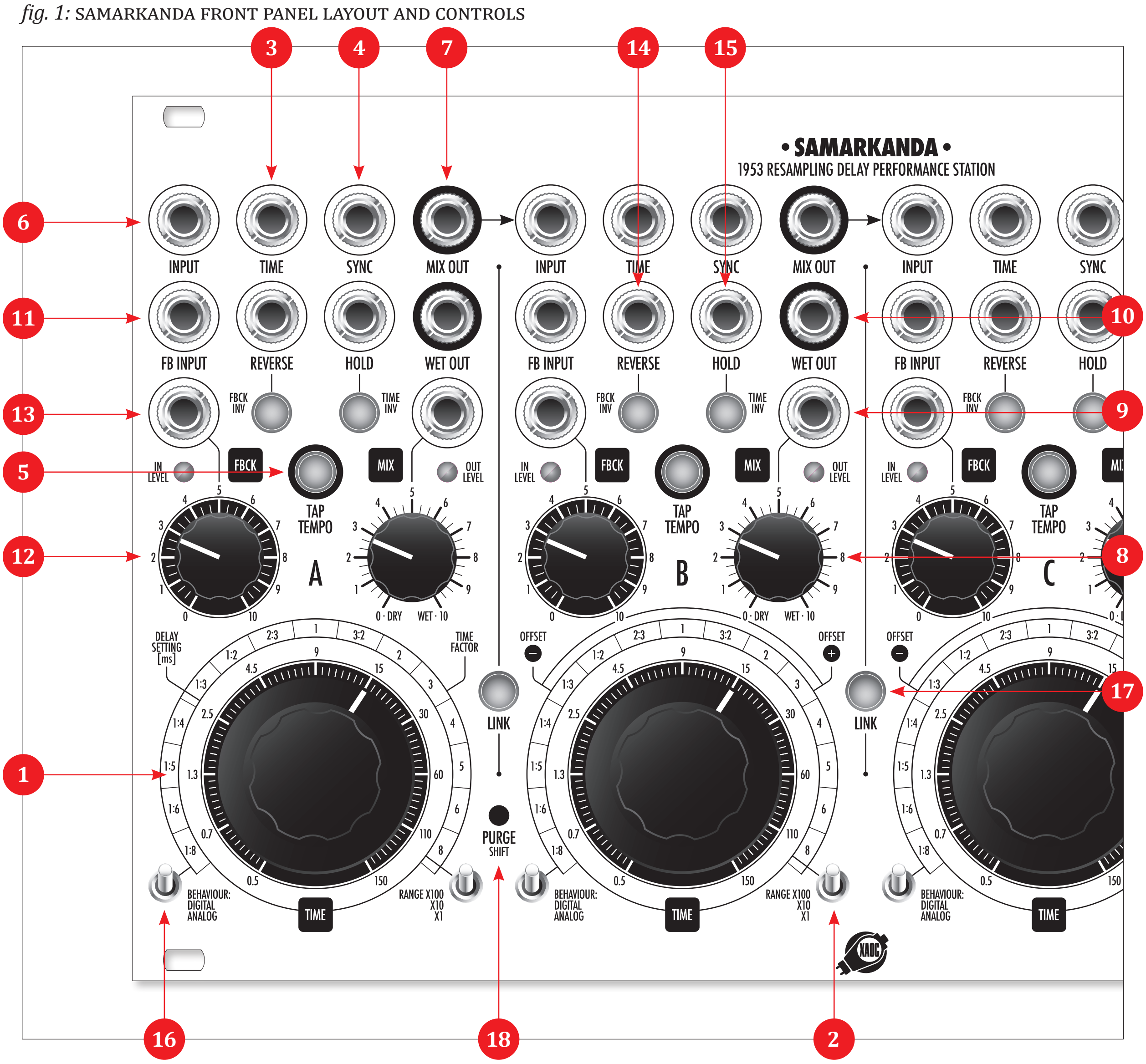
The return input of the feedback loop ⓫ is labeled **FB INPUT**. The amount of feedback is controlled by the **FBCK** knob ⓬ and its associated CV input ⓭, which accepts bipolar voltages. Please keep in mind that the **FB INPUT** is digitally normalled to the **WET OUT**, which means there is a clean, lossless feedback loop if no cable is patched.

Each section of the panel features two additional performance buttons with associated jacks that accept gate or trigger signals (in both, the rising edge flips the state of the button). The **REVERSE** button and gate/trigger input ⓮ select between a traditional delay effect and the "reverse delay" effect, in which segments of the signal from the delay buffer are played backwards. The **HOLD** function ⓯ instantly freezes the contents of the delay buffer and loops it, ignoring the input signal.

The **BEHAVIOUR** switch ⓰ located to the left of the big dial selects between two fundamentally different methods of handling delay time changes. The **DIGITAL** option employs a semi-granular method by crossfading between varying positions of the delay buffer. Thus, there is no pitch change when the delay time is modified. The **ANALOG** option uses a variable sampling rate yielding realistic pitch changes that resemble classic analog tape and BBD delays.

The **LINK** buttons ⓱ between each neighboring pair of panel sections control the combining of channels into linked pairs or larger groups. There are multiple combinations in which channels may

fig. 1: SAMARKANDA FRONT PANEL LAYOUT AND CONTROLS



be linked, which will be discussed in a separate section of this user manual. In general, the manual and CV-controlled parameters of channels linked to some other (master) channel are relative. Parameter changes in a master channel are followed by the linked channels. Observe the additional **OFFSET–** *and* **OFFSET+** *marks around the* **TIME** *knob scales in channels* **B**, **C**, *and* **D**, *which remind you about these alternative functions.*

The singular **PURGE·SHIFT** *button* ⓲ *allows you to clear the contents of all delay buffers in all channels (thus removing any signal buildup) and start from scratch. The function is executed upon releasing the button after being momentarily held. This button is also used for activating options when combined with other buttons: inverting the polarity of the feedback (by pressing the* **REVERSE** *button while holding* **SHIFT**), *as well as flipping the direction of CV that modulates the delay time (by pressing the* **HOLD** *button while holding* **SHIFT**). *Note the additional labels printed near these buttons, which remind you about their secondary functions.*

Samarkanda maintains the state of all function buttons and recalls them upon a power cycle.

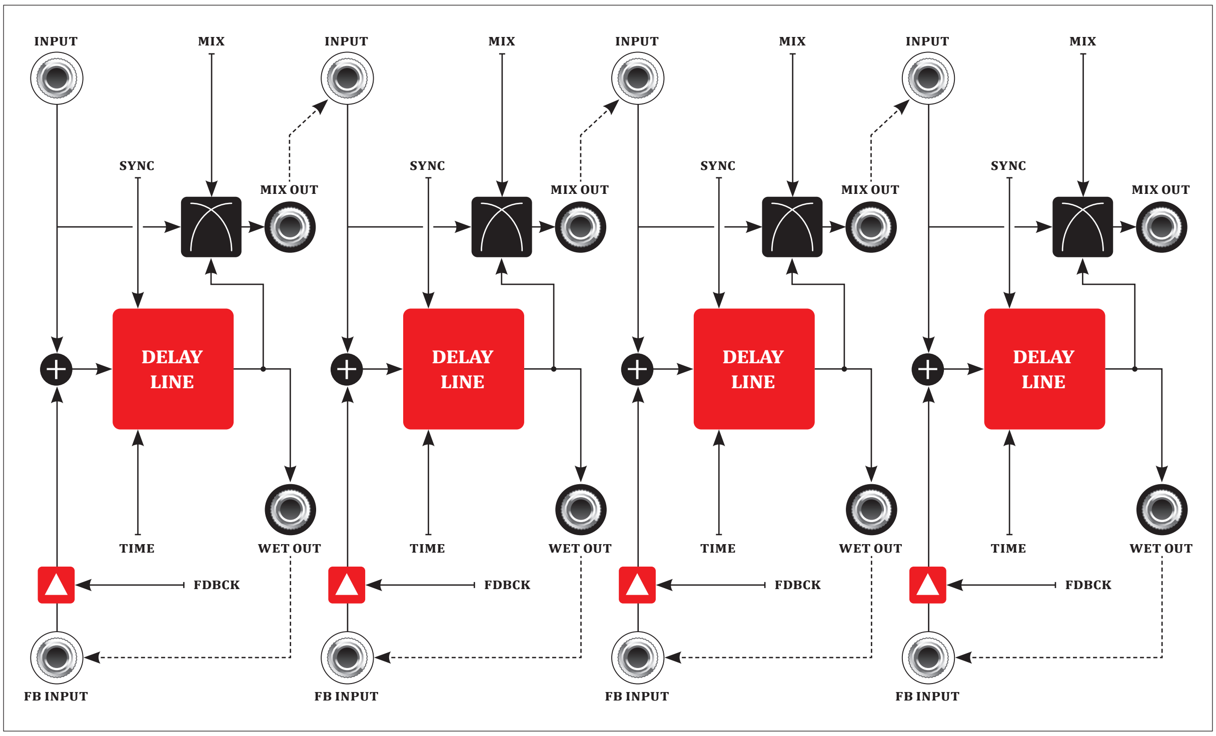
DELAY INTO DELAY INTO DELAY INTO DELAY

Figure 2 shows the block diagram of the device. Each of the four channels uses a separate delay line, which can be adjusted and modulated independently. In general, a bank of delay effects can be combined in several ways, e.g., in parallel or in series (stacked), which yields very different sonic results. By default, the channels in Samarkanda are stacked; this means that the mixed signal from the first channel is passed as the input to the next channel, and so on.

When four delay effects with feedback are stacked, each with a different delay time, they generate a highly complex pattern of echoes. With each added delay, the overlapping patterns become increasingly dense and unpredictable, creating intricate rhythmic and tonal interactions.

When mixing a delayed signal with the original, a natural effect called comb filtering occurs, where certain frequencies cancel out due to phase interactions. This effect becomes more complex with four stacked delay channels as multiple comb patterns overlap. As a result, when processing long, sustained sounds (e.g., drones), you may notice a natural reduction in perceived loudness that is

fig. 2: BLOCK DIAGRAM OF SAMARKANDA



not a fault but an inherent characteristic of delay-based processing that we encourage you to explore creatively in your sound design.

Besides this default arrangement, each channel of Samarkanda can be used differently and with diverse scenarios of synchronized or jointly operated groups, making the device a multi-faceted, flexible, and highly performative instrument.

fig. 3: THE PRINCIPLE OF ANALOG DELAY EFFECT WITH VARIABLE OPERATING RATE

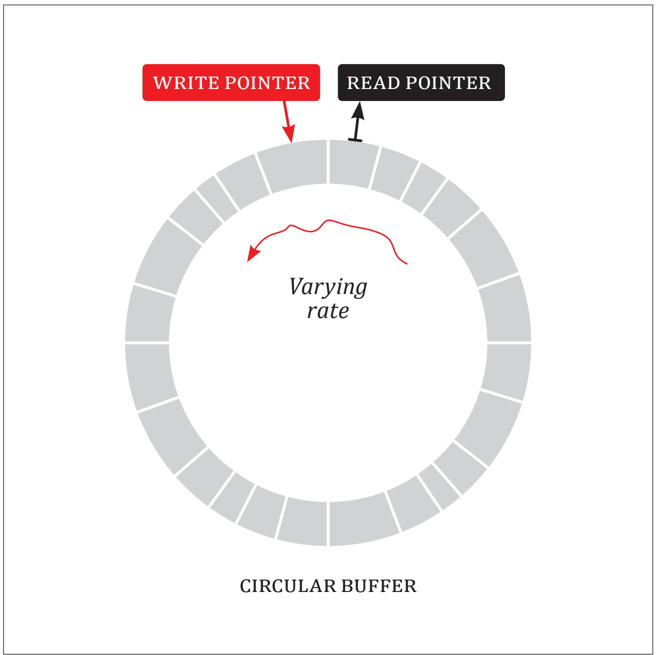
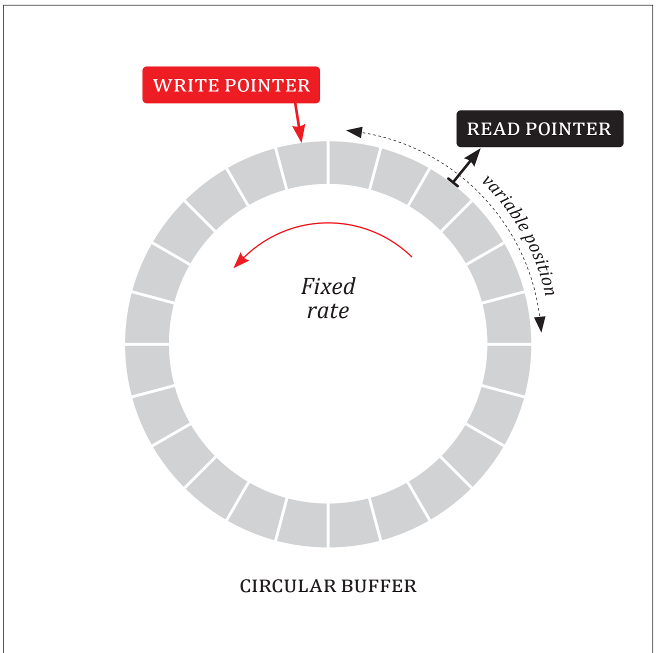


fig. 4: THE PRINCIPLE OF DIGITAL DELAY EFFECT WITH MOVING READ POINTER



ANALOG VS DIGITAL BEHAVIOUR

How a delay effect device responds to delay time changes (set manually or by CV) is crucial to the overall experience and sound. Samarkanda offers two fundamentally different algorithms: emulated analog (BBD/tape-like) and digital (granular-like). Consider that most analog delays operate using a fixed-capacity buffer (e.g., a loop of magnetic tape or a series of BBD chips), and varying the delay time is achieved by varying the rate at which the buffer operates (varying the speed of magnetic tape or varying the clock of the BBD chips). Changing the rate on the fly causes all current contents of the buffer to be squeezed or stretched in time (fig. 3), which pitches the output signal up or down until the old contents of the buffer are replaced by a new portion of the incoming sound that is written at the new rate.

The "analog behaviour" in Samarkanda offers precisely these types of effects. It creates on the fly a flexible signal model (similar to vector graphics) that can be resampled with extreme factors. Thus, the output signal may be played back at a varying rate, significantly different from the recording one.

Note that the change of pitch is exactly tied to the ratio of the read and write speed of a particular portion of sound. That is why when the delay time is rapidly switched between two integer clock divisions, it results in a harmonic ratio (pure intervals in just intonation), contributing to Samarkanda's signature sound (similar to Xaoc Devices Sarajewo).

Typical digital delays employ a variable length buffer, usually a digital memory freely accessible by write and read pointers (fig. 4), which operate at a fixed sampling rate. The delay time depends on the distance between these pointers, and changing the time involves moving the read pointer closer to or further from the write pointer, which may cause glitchy discontinuities of the signal.

One way of avoiding glitches is to "catch up" by reading the part of the signal to be skipped at a faster or slower rate, thus yielding audible pitch changes proportional to the speed of changing the delay time (acceleration or slowing down). Note *that this change of pitch occurs only during the change of the delay time (and not after the change is done), and it is not the same as the distinctive analog effect described above.*

The method implemented in Samarkanda is to smoothly crossfade between consecutive positions of the reading pointer, which does not produce changes in pitch. Instead, there is a granular-like effect of "scrubbing" the buffer. We use adaptive crossfading that considers the current delay time and its acceleration.

Switching between the two behaviour options is possible at any time, in each channel separately, by operating the **BEHAVIOUR** *switch. During normal delay operation (i.e., with the* **HOLD** *function off), this yields seamless switching and can be used performatively without losing signal integrity. When* **HOLD** *is active, the switch selects between two entirely different ways the loop is operated (see subsequent section), and the audible results may change significantly.*

FEEDBACK CONTROL

Feedback is an essential feature of every delay effect. The control of feedback amount in Samarkanda is done through the **FBCK** *knob and a bipolar CV which is added to the position of the potentiometer. A full turn is equivalent to a CV change by 10V. For optimum experience, these two are subject to a nonlinear control response starting sharply from 0% and reaching values above 80% very quickly, devoting nearly half of the range to the last few % and getting very close to 100%.*

When the delayed signal is mixed with the new incoming signal and returned to the delay line, it yields a cascade of echoes. That also means the signal energy at the input is potentially increased by adding multiple copies of itself, especially when the delay time is short and the sounds are long and sustained. To prevent distortion in Samarkanda, the input signal is gradually attenuated when feedback exceeds 50%. Furthermore, two methods are used to handle situations when the summed signal becomes very hot:

- *with* **ANALOG** *selected, a carefully modeled soft clipping is applied to signal peaks to emulate the saturation effects of analog delay devices;*
- *with* **DIGITAL**, *a dynamic compressor is applied to mitigate the amplitude increase. This compressor has been designed to be unobtrusive and transparent.*

To achieve infinite and non-decaying echoes, engage the **HOLD** *function, which prevents new signal from being added to the loop. You can still exercise the "sound on sound" technique by setting* **FBCK** *to maximum and disengaging* **HOLD** *only for the periods you want new sounds added to the loop.*

The **FB INPUT** *and* **WET** *jacks are provided for creating external feedback loops with additional processing, such as filtering, modulation, and distortion. Note that with four channels of Samarkanda, there are plenty of options to create loops within and across channels, which may yield a complex net of interconnections resembling those of the FDN (feedback delay network) reverberation techniques.*

Please keep in mind that external feedback loops increase the risk of distortion if any of the additional processing elements introduce gain. The **FBCK** *potentiometer must be operated carefully to counteract the uncontrolled buildup of signal energy.*

DIGITAL FEEDBACK AND STACKING CHANNELS

If no source signal is patched to the input of the next channel, the digital **MIX** *signal data of the previous channel is internally passed to the next channel, bypassing the D/A and A/D conversion. This virtual normalization enables lossless cascaded processing of signals in a stack of delay effects. Patching an external audio signal into the input breaks the connection. For example, stereo processing of a pair of signals, or two copies of the same signal, requires using these inputs separately.*

FB INPUT *jacks apply the same principle. With no external signal patched in (for example, a processed signal from an external loop), a digital copy of the delayed signal from the same channel is used for the feedback. Please remember that this smart normalization uses a cable detection mechanism that shuts down unwanted signals when their source should be ignored. Therefore, for optimum results, long cables that are not connected on their other side should not be patched to Samarkanda's inputs.*

CONTINUOUS, SYNCHRONOUS, AND TAP-TEMPO OPERATION

Each channel of Samarkanda (when it is not linked to a master channel, see next section) may operate with continuous time control, synced to an external clock, or with tempo defined by the **TAP TEMPO** *button.*

With continuous operation (button blinking green), the big **TIME** *knob allows you to dial any value between 0.5ms and 150ms with the* **RANGE** *switch set to X1, between 5ms and 1500ms at the X10 position, and between 50ms and 15s when set to X100. While the three ranges overlap, their endpoints can be extended by adding external CV; however, the 0.5ms and 15s limits cannot be exceeded in any situation.*

Patching a low-frequency clock signal into the **SYNC** *jack automatically switches that particular Samarkanda channel to synchronous operation, which is indicated by the* **TAP TEMPO** *button blinking red. The period of the clock is precisely translated to the delay time with the inclusion of a division or multiplication factor from the list of 1:8, 1:6, 1:5, 1:4, 1:3, 1:2, 2:3, 1, 3:2, 2, 3, 4, 5, 6 and 8, selected by the big* **TIME** *knob. The position of the* **RANGE** *switch is ignored.*

The control voltage patched into the **TIME** *input adds to the position of the* **TIME** *knob and modulates the division factor. The incoming clock period is continuously monitored, and its fluctuations are followed. Note that two subsequent clock pulses are needed to update the information. Keep in mind that Samarkanda will not sync properly to a clock, which, after applying the factor set, would yield a delay time out of the 0.5ms to 15s range. To return to continuous operation, unplug the clock source.*

Tap-tempo is yet another way of defining the delay time. It requires that no cable is plugged into the **SYNC** *jack. Pressing the button at least two times sets the time base to the measured span between these presses, which is indicated by blinking in yellow. The actual delay time is this time base modified by the factor set with the* **TIME** *knob and CV, just as with external clocking. The position of the* **RANGE** *switch is ignored. To return to continuous operation, long-press the* **TAP TEMPO** *button until it starts blinking green again.*

LINKING CHANNELS

Two or more channels of Samarkanda can be linked so that they may be operated jointly using the controls of the master channel (the leftmost channel of the linked group). There still may be some differences between channels set by the controls and modulation CV inputs of the secondary channels. This option allows you to easily configure stereo pairs and create strictly synchronized polyrhythmic groups, chorusing, pairs slowly phasing out, or detuned clusters of resonators.

To link two channels, press the **LINK** *button between the corresponding sections of the front panel. At any time, you can extend the group by linking another channel or split a group by pressing the button again. There are eight possible configurations, among them two independent pairs of linked channels:* **A+B** *and* **C+D**. *In any combination, the leftmost channel of the group is the master channel.*

When a channel is linked to some other (master) channel, its delay time is set relative to that channel. The **TIME** *dial acts as a bipolar control with the 1:1 point in the middle position. Turning it to the left of this point shortens the delay, and turning to the right increases it. Similarly to the unlinked state, the linked channel can be switched between continuous or synced operation by a single press of the* **TAP TEMPO** *button*

The green backlight of this button indicates continuous adjustment of the delay time with respect to the master channel, available in 3 ranges selected by the **RANGE** *switch:* **X100** *means the time can be adjusted in full 1:300 scale,* **X10** *is a significantly reduced scale for finer adjustments, and* **X1** *selects a very narrow range around 1:1. The chosen* **RANGE** *setting determines both the scale of the TIME dial as well as CV patched to the* **TIME** *jack.*

The red backlight of the button indicates discrete divisions/multiplications set between 1:8 and 8:1, exactly as when a channel is synced to an external clock. Keep in mind that these are division or multiplication factors applied to the delay time set in the master channel, which is opposite to clock rate division and multiplication. If the master channel is also synced to an external clock with its own factor, the combination of multiplications and divisions may result in unusual time signatures, like

5/6, 2/9, etc. Note that the **RANGE** *switch and the* **SYNC** *jacks are ignored.*

Apart from the **TIME** *dials, also* **FBCK** *and* **MIX** *knobs work relatively. In the middle position, the parameters reflect the settings of their counterparts in the master channel. The potentiometers' range is extended in order to allow them to fully counteract the position of the master channel controls.*

For example, if the **FBCK** *knob in the master channel is set to 10, turning the* **FBCK** *knob in the linked channel fully CCW reads as "-10", so the resulting feedback amount is 0. Similarly, if the* **MIX** *in the master channel is set to full* **WET***, turning the* **MIX** *knob in the linked channel down to 2.5 reads as "-5" (remember about the scaling!), so the resulting output signal available at the linked channel's* **MIX OUT** *will contain 50% of the wet signal. Remember that you can link channels that are processing different signals, so many combinations of the wet/mix signal proportions are possible.*

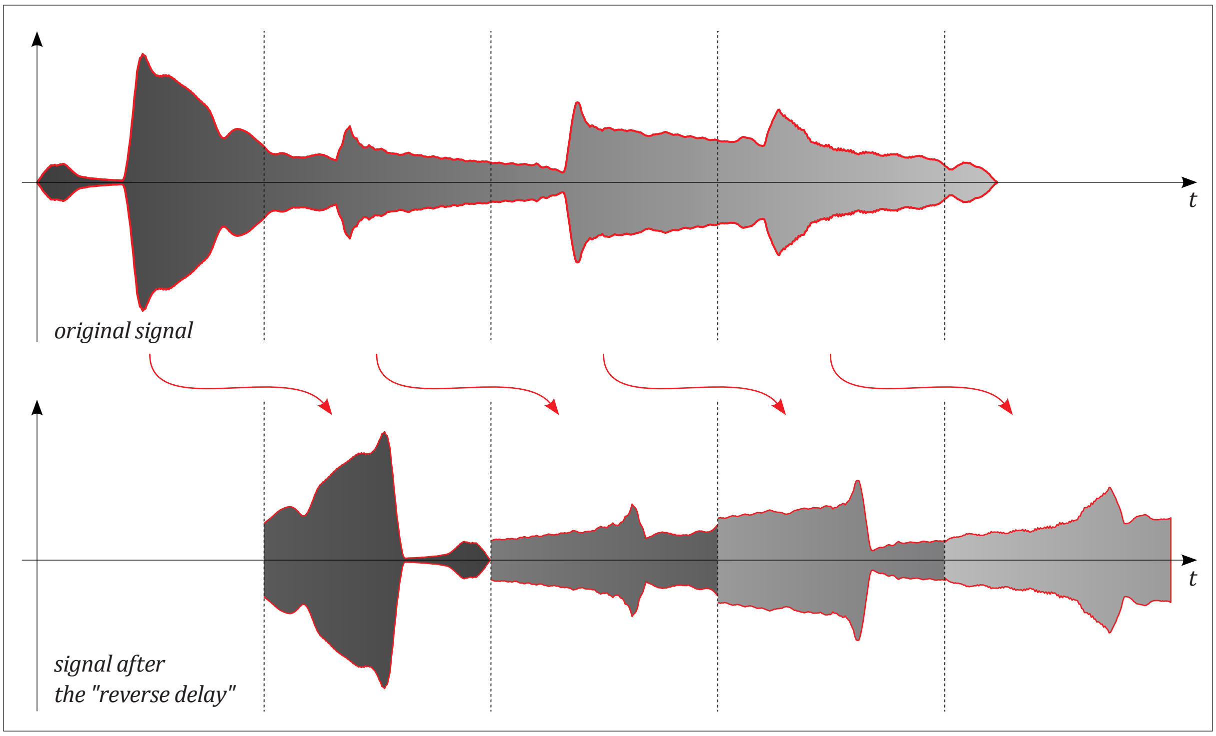
REVERSE DELAY

Each channel of Samarkanda may act as a "reverse delay." It is obviously a misnomer, as it is impossible to un-delay sounds that are already late. Instead, a portion of the signal in the buffer is played backwards by moving the write and read pointers in opposite directions until the end of the buffer is reached, and the process repeats. Note that this means there is a periodic discontinuity when the read pointer reaches the oldest sample of the signal available in the buffer and jumps back to reading the newest sample, which has just been written. Additional short fades mask this jump to avoid audible glitches.

Adding feedback to the "reverse delay" effect applies this reversing process to the already reversed sound, effectively canceling it in the second repetition. That is why each second echo is not reversed, which may be undesirable. The solution is to use two stacked delay channels: the first one with **REVERSE** *engaged and no feedback, full wet, and the second one without* **REVERSE** *but with the feedback up.*

Note that the reverse effect is most pronounced for long delay times because, for longer segments of sounds, there is a better chance of the listener hearing their temporal structure being played backwards. For very short delay times, the audible difference between reversed and nonreversed effects is dominated by the cyclic fades at the edges of the buffer, making it less attractive.

fig. 5: THE EFFECTS OF REVERSE DELAY



HOLD AND LOOPING

Engaging the **HOLD** *function in any channel (either by pressing the button or via the trigger/gate input) creates a loop: the content of the buffer is frozen and is not overwritten by the input signal but is being played back in a cycle. The length of the loop is equal to the delay time that was set in the instant of activating* **HOLD**. *However, during the cyclic playback, the length can be modified by operating the* **TIME** *dial, the* **RANGE** *switch, and applying CV.*

When **DIGITAL** *is selected with the* **BEHAVIOUR** *switch, the length of the cycle is modified by selecting a shorter or longer segment of the whole buffer, which keeps the recent 15 seconds of the signal captured just before engaging* **HOLD**. *By operating the* **TIME** *controls and the* **RANGE** *switch, you can freely sweep it within the 0.5ms to 15s limits, keeping the content of the loop intact.*

When the **BEHAVIOUR** *switch is in the* **ANALOG** *position, the content of the looped segment is fixed. Still, it is being resampled on the fly so that, depending on the controls, it speeds up or slows down (with associated pitch changes). The* **TIME** *controls (knob and CV) allow you to squeeze or stretch the contents of the loop in a very wide range.*

Keep in mind that you should not expect any reasonable audio effects of a loop that is compressed from many seconds down to milliseconds, or stretched from milliseconds to multiple seconds, as all audible frequencies are then shifted out of the range of hearing. Strange artifacts at extreme settings are normal side effects of resampling going too far.

When the **HOLD** *function is activated with* **ANALOG** *selected, the stretching/pitch effects may be used to "play" the sound caught in the buffer using the* **TIME** *CV input. Setting the response to +1V/octave with the* **PURGE** *and* **HOLD** *buttons (see the section below) turns Samarkanda into a crude sampler.*

Depending on the initial setting of the **TIME** *knob, not only longer sounds and phrases but also single-cycle and multiple-cycle waves can be sampled accurately with the help of the* **SYNC** *input. However, please note that the buffer is continuously being played back in a loop, and there is no way to synchronize the endpoints of the looped phrases to any external signal.*

At any time, either with **ANALOG** *or* **DIGITAL** *selected, you can return to the original looped material without any signal loss by resetting the* **TIME** *settings to values active at the moment of* **HOLD** *activation.*

THE PURGE/SHIFT BUTTON

Pressing and holding the **PURGE·SHIFT** *button prepares Samarkanda to clear the contents of all delay buffers in all channels. This function is convenient in situations of uncontrolled signal level increase due to heavy feedback. Upon releasing the button, all four delay buffers are cleared, and writing to them begins anew.*

However, the clearing action is not performed if you press and hold the **PURGE·SHIFT** *button and then press any channel's* **REVERSE** *or* **HOLD** *button. You may press more than one and observe their illumination changing to reflect the options activated in respective channels. Changes are effective upon releasing* **PURGE·SHIFT**.

The **SHIFT + REVERSE** *button combo flips the polarity of the feedback signal in each channel between positive (the factory default) and negative. The audible effect is subtle for long delay times but becomes pretty dramatic for very short times (e.g., in resonator applications) because it affects the locations of peaks and notches in the frequency domain.*

The **SHIFT + HOLD** *combo flips the polarity of the CV input between (the factory default) -1V/octave and +1V/octave; this is handy when you want to play the captured loops with a keyboard or sequencer. Note that only the polarity of the CV is inverted, not the direction of the big* **TIME** *controls.*

SIGNAL LEVEL INDICATORS

Each channel of Samarkanda is equipped with two multi-color LED indicators of the analog **INPUT** *and* **MIX OUT** *signal levels. The indicators employ a PPM (peak detection) method with an appropriate discharge time constant, allowing for easier observation of short transients.*

Levels from silence up to the common Eurorack 10Vpp are shown by the LEDs gradually lighting *green. Exceeding 10Vpp is indicated by the color turning yellow, and finally, red whenever very hot levels (above 16Vpp) are reached. Note that the Samarkanda ADCs and DACs can handle signals up to 20Vpp, and the red color only warns of a hot signal, which engages the internal soft saturation.*

Please remember that the indicators show only levels of analog signals and not the digital signals, which are passed from channel to channel due to virtual normalization. Therefore, the input LEDs will be unlit when no cable is plugged into the **IN** *jacks.*

FIRMWARE UPDATES

To make firmware updates easy, Samarkanda is equipped with a mini-USB connector at the back. It should work with any computer and operating system supporting USB storage. It does not require any dedicated software; it just creates a virtual drive where you need to place the firmware file and reboot the module. Each firmware package will come with instructions on how to perform the update.

PATCH EXAMPLES

• TWO STEREO DELAYS: *To configure Samarkanda as two independent stereo delay effects, link channels* **A+B** *and* **C+D**, *set all knobs in channels* **B** *and* **D** *to the middle position, and set the* **RANGE** *and* **BEHAVIOUR** *switches in channels* **B** *and* **D** *the same way as* **A** *and* **C**. *Delay time, feedback, and mix controls from channels* **A** *and* **C** *will affect their second channels consistently.*

• PING-PONG DELAY: *Link two channels and cross-couple their feedback loops. Patch the* **WET OUT** *signal from the master channel to the* **FB INPUT** *of the secondary channel, and vice versa. Set the* **FBCK** *and* **MIX** *controls in the secondary channel to the middle position. Experiment with non-centered settings of the* **TIME** *dial in the second channel for various echo patterns. Don't forget to hard-pan Samarkanda's outputs in your mixer.*

• MAXIMUM DELAY TIME: *To achieve one very long delay line (up to one minute of delay), use the default stacked configuration – set all* **RANGE** *switches to X100, all* **MIX** *knobs to 100% wet, and* **FBCK** *knobs in channels* **B**, **C**, *and* **D** *to fully CCW. Patch a cable from the* **WET OUT** *of channel* **D** *to* **FB INPUT** *of channel* **A**. *Wet/dry mixing must be done externally (e.g., by using Samarkanda in a mixer aux loop).*

• BOUNCING A LOOP: *While playing a loop, or just any interesting sound in channels* **A**, **B**, *or* **C**, *you can easily make an exact copy of this loop in the next channel by using normalization. Link the next channel to the current one, set the second channel's* **FBCK** *to 0,* **TIME** *to center for 1:1, and ensure it is operating in sync (***TAP TEMPO** *button blinking red). Leave the second channel's* **HOLD** *button off for at least the length of the loop, then turn it back on.*

• TUNED RESONATOR: *Set the* **BEHAVIOUR** *switch to* **ANALOG**, *and the* **RANGE** *switch to X1. Set the* **FBCK** *knob to near max. Use the continuous time control with pitch CV patched into the* **TIME** *jack. Set sensitivity to +1V/octave using the* **PURGE** *and* **HOLD** *combination. This patch offers particularly interesting effects when excited with complex impulses, for example, those offered by Xaoc Devices Zadar in* BANKS **S, U, V, W,** *and others.*

• REAL-TIME GRANULATOR: *Use any channel with* **BEHAVIOUR** *set to* **DIGITAL**, **RANGE** *to* **X10**, **FBCK** *to 0,* **MIX** *set to fully wet. With continuous time control (LED blinking green), patch some random modulation source to the* **TIME** *CV jack. To mimic higher density grain, increase feedback or feed the output signal to another Samarkanda channel. Note: The result strongly depends on the character of your modulation.*

• REVERB: *You can combine all four channels into an FDN reverb-like structure by adding a matrix mixer. Patch four* **WET OUT** *signals into the mixer inputs and use its four outputs to return to four* **FB INPUT** *jacks. Split the original audio into four copies and patch them into all Samarkanda's* **INPUT** *jacks. Use any two* **MIX OUT** *jacks as the stereo output pair. Remember that achieving a reverb effect relies mainly on short delay times with deep feedback. Therefore, you must watch out for feedback levels, as it is easy to introduce heavy distortion.*

ACCESSORY

Our Coal Mine black panels are available for all Xaoc Devices modules. Sold separately. Ask your favorite retailer. •

TECHNICAL SPECIFICATION

WIDTH	DEPTH TOTAL	CURRENT DRAW	REV. POWER PROTECT.
42hp	*40mm (incl. ribbon bracket)*	*+210mA* / *-85mA*	*protected*

INPUTS		OUTPUTS	
INPUT	*0 to 20Vpp, recommended 10Vpp*	**MIX OUT**	*0 to 20Vpp*
FB INPUT		**WET OUT**	
TIME	*-5V to +10V*		
SYNC	*any clock signal, rec. 5V*		
REVERSE	*+5V standard trigger or gate*		
HOLD			
FBCK	*-10V to +10V*		
MIX			

FREQUENCY RANGE
16 Hz to 20 kHz

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