ELECTRO-ACOUSTIC RESEARCH

Operation Manual: Model 41 Steiner 4P Filter

Contents

Introduction

In June of 1992, Nyle Steiner designed an updated version of the filter found in his SynthaSystem, a revision that was never fully completed, and intended for his EVI. In a discussion with Steiner in 2008, the EAR Model 41 -based on this design -was slated as a limited edition module in 2009 to celebrate the 5 year anniversary of Plan B. As with other modules planned for that year, the M41 was never released.

In Jan of 2014 the collaboration resumed with a few design additions suggested to him: IFM SYNC (the 'twin' function found in the M11), a HYPER Q Regeneration function, a +/- bipolar VC input, VC resonance, along with all benchmarks true to the original 4P Filter design including a variable intensity control (Spectral Energy Distribution), a four pole 18-stage diode string for 24dB/ oct response, oscillation bandwidth of .5hZ to 16kHz suitable for LFO or VCO applications, user-scalable 1VOLT/OCT tracking, and full incorporation of integrated verses discrete transistor amplification architecture.

Renamed the Steiner 4P Filter, the Electro-Acoustic Research Model 41 available to market is the third complete revision of the ordinal circuit, all of which involved hands-on qualification by both Peter Grenader and Nyle Steiner over an eight month period in 2014. The production M41s are being made available thorough an exclusive manufacturing/distribution arrangement with Darkplace Manufacturing in Portland, OR. USA.

From Peter Grenader:

On behalf of Nyle and myself, we hope you appreciate one of the best kept secrets of modular synthesis in the past 23 years - and a new generation of Steiner instruments. Happy wiggling!

From Nyle Steiner

The Model 41 was derived from a new filter design which i started using during my studio session work with the EVI in 1992. It was first used for the Columbo and Matlock television series sessions. It's great that this design has now been made available to the public. I hope you enjoy it.

The Fundementals of Subtractive Synthesis: Filters

History

The development of the modern day active electronic filter was introduced in the 1940s as a means of decoding specific frequency bands from larger frequency bandwidth signals, without the use of inductors incorporated in passive filters. They were largely used within radio stations and when early after-hour attempts in experimental music began in these radio studios in the mid 1950s, filters were fundamental in expanding the timbral vocabulary of synthesized electronic sound by what's come to be known as subtractive synthesis. One of the earliest known pieces utilizing subtractive process via filters was Karlheinz Stockhausen's *'Gasang der Junglinge' (Song of the Youths)* in 1956.

By the 1960s, advancements in semiconductor technology - namely the transistor - afforded instrument designers and by proxy, composers greater precision through the use of external voltages to vary filter center frequency via a technique known as voltage control - the cornerstone of modern-day analog modular music synthesizers.

Types and Parameters of Filters for use in Electronic Music

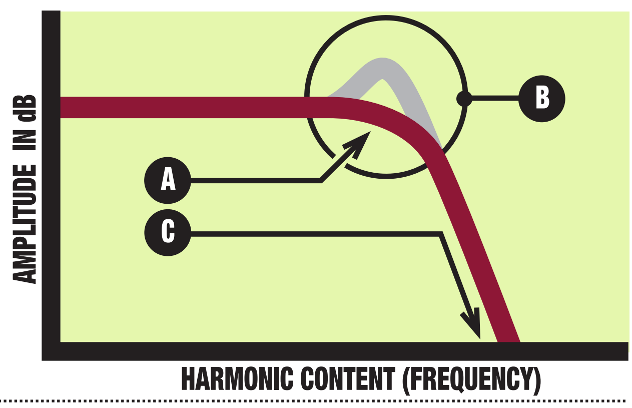
There are three primary types of filters utilized in Electronic Music - Lowpass, Bandpass and Highpass, with other modes (Allpass, Notch) derived from a combination of these three. For the purpose of this manual, we are going to focus on the three primary modes, all of which are available and voltage controllable in the EAR Model 41 Steiner 4P Filter.

A **Lowpass Filter** passes harmonics with a frequency lower than a cutoff set by the FREQUENCY CONTROL and attenuates harmonics within the bandwidth higher than that set frequency. A **Highpass Filter** works in inverse, allowing harmonics higher than the set cutoff frequency to pass through while attenuating harmonics lower than the set cutoff. A **Bandpass Filter** allows frequencies between a set (fixed or variable) range to pass, while attenuating those outside of those limits. All three modes have a specific effect in electronic music applications with the sonic character of each determined by three basic parameters: **A** Cut Off Frequency **B** Factor (Resonance) **C** utoff Slope.

FIGURE 1 • Primary Filter Functions

This graph plots the effect of a LOWPASS Filter on the audio spectrum where the cutoff determines which frequencies pass through and which are attenuated, and the severity of that cutoff.

On a variable filter, the horizontal position of the elbow **(A)** is determined by the **Frequency Control.** The two contours shown at that elbow **(B)** indicate the differences of low (red) and high (gray) Q Factor as determined by the **Resonance Control**. **(C)** indicates the angle of the Cutoff Slope, measured in dB/octave.



Filter Fundementals (cont'd)

Types and architectural considerations of filters used in Electronic Music

1963 marked a pivotable year in the development of electronic music with the arrival of modular analog synthesizers with Voltage Controlled Filters, along with VCAs being the primary components used for subtractive synthesis. There were two basic types of cores utilized in the many different filters which were released over the next 15 years: those having a State Variable engine, and those using a Sallen and Key. Unlike the Plan B Models 11 and 12 which are State Variable - The EAR Model 41, like it's predecessor the Steiner Synthasystem filter, like the Plan B Model 13 filter incorporate Sallen and Key topology.

Due to the architectural simularities of the M41 and Synthasystem cores, the two filters retain much of the same persnality traits with some significant differences. Apart from incorporating intergrated (IC) vs. discrete (transistors) unity gain stages - the M41's 24dB/octave cutoff slope is about twice as steep as it's process. The measurement of dB/Octave refers to the slope determined by the rate of decline in amplitude (dB) per octave (harmonics) plotted logarithmically from the cutoff frequency, with steeper response yielding sharper/more aggressive timbral influence.

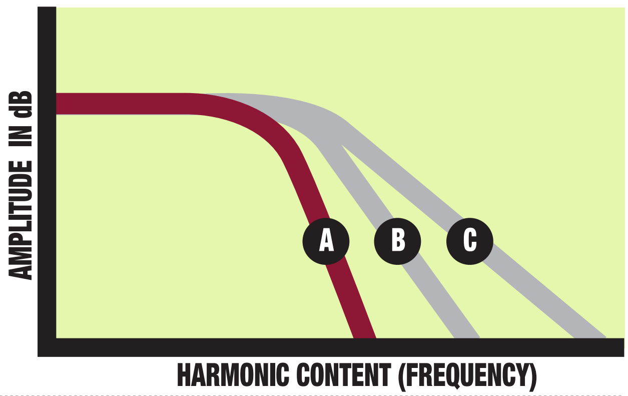
FIGURE 2 • Comparative Cutoff Slopes (dB/oct)

This graph plots the relative cutoff slope of on a logarithmic scale

(A) 24db/Octave

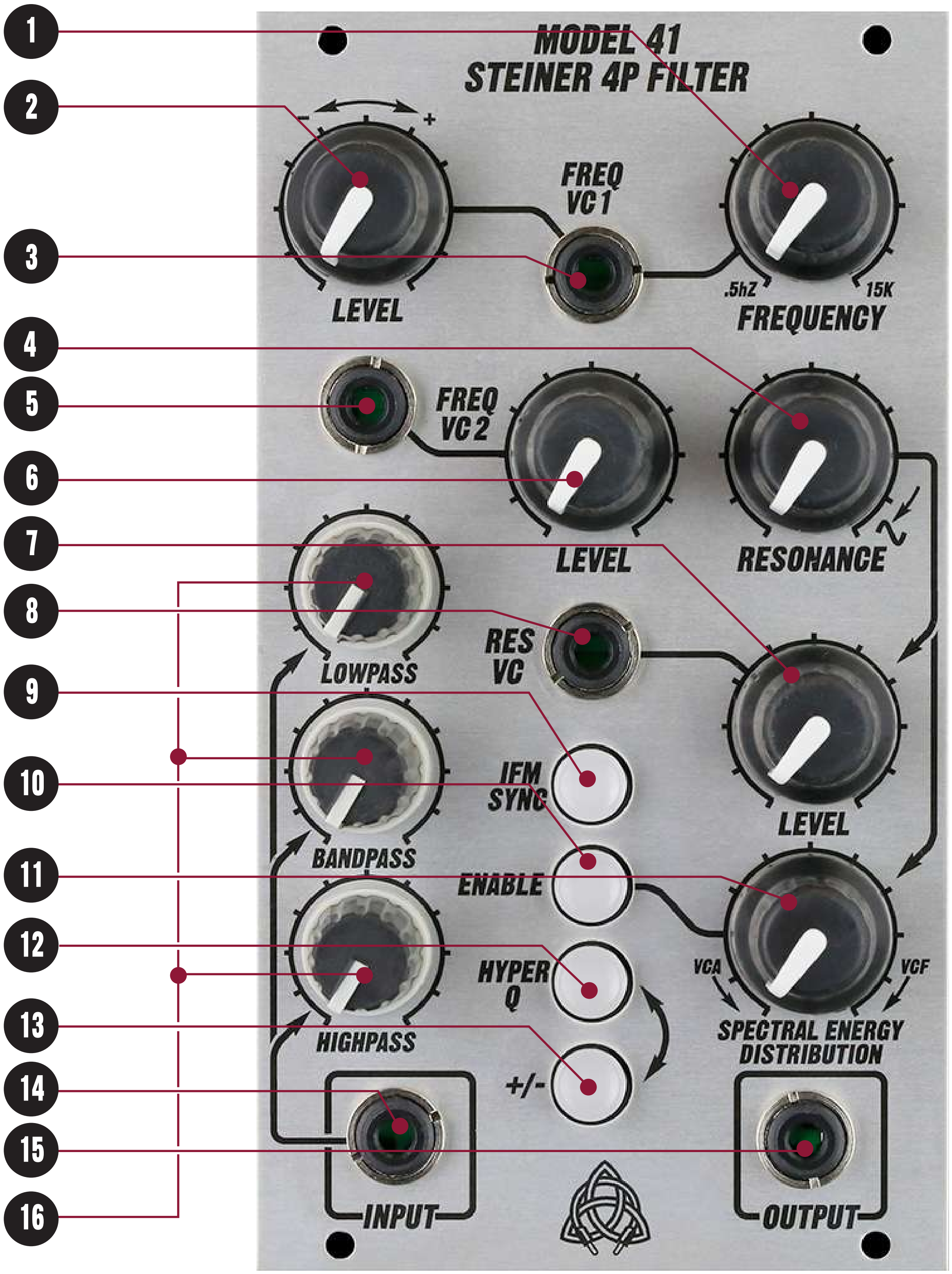
(B) 18dB/Octave

(C) 12dB/Octave



Further improvement has been introduced on the M41 by the Spectral Energy Distribution control and two preset distortions modes (IFM Sync and Hyper Q), all three unique to EAR Filters. In depth description of these three functions will be outlined in chapter four: Advanced Filtering Techniques , page 10.

Summary of Controls: EAR Model 41 Steiner 4P Filter



Summary of Controls (cont'd)

1	Frequency manual control (range .5hZ to 15kZ)
2	Bipolar (+/-) Frequency modulation input 1 attenuator
3	Frequency modulation voltage control input 1 (External input for no. 2 above)
4	Filter Resonance manual control
5	Frequency modulation voltage control input 2 (External input for no. 6 below)
6	Frequency modulation input no 2 Std attenuator - 1v/oct manual adjust
7	Filter Resonance Voltage Control input standard attenuator
8	Filter Resonance Voltage Control Input
9	IFM Sync Frequency Modulation mode (engaged when lit)
10	Spectral Energy Distribution Enable (engaged when lit)
11	Spectral Energy Distribution control
12	Hyper Q Resonance Modulation mode (engaged when lit)
13	Hyper Q Resonance Modulation polarity (-/+)
14	Signal Input
15	Signal Output
16	Output Mode Level Attenuators (Lowpass, Band Pass, Highpass)

Connecting and Mounting your EAR Model 41

Like all Plan B/EAR modular analog instruments, the Model 41 adheres to the mechanical and electronic conventions of Require founder Doepfer MusikElektronok requirements as outlined on http://www.doepfer.de/a100_man/a100m_e.htm Electrical and signal parameters are covered in detail in chapter five: Specifications. In the meantime, you probably want to plug in your EAR Model 41 and take for a test drive. As in all EAR products, power is connected via either a 16 or 10 pin dual in line header connector which comes with your module when purchased. There are a few guidelines to do this correctly however. While all EAR products include protection circuitry to assure no damage will occur if a module is not connected properly, EAR cannot be responsible for damage to other modules which may occur if it's not, so it's best to take a few seconds to make sure it's done correctly.

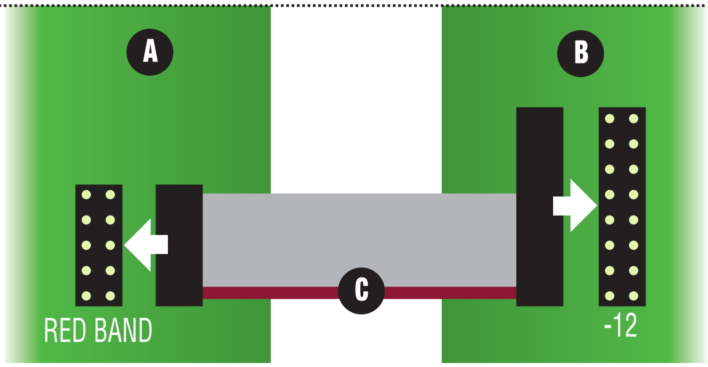
To connect power to your Model 41, the flat cable will need to be properly installed, On one end there is a smaller 10 pin DIP (Dual in Line) header connector, on the other a 16 pin DIP header connector. An important feature to note is that the 10 wire flat cable is distinguished by a red band on wire (pin 1). As mentioned, there are two different connectors on each end of this cable: The 10 pin end is designed to be connected to the module, and the other 16 pin end designed for connection to your power source - which for the purpose of this manual will be limited to the Doepfer power backplane, although alternate sources are available. Please refer to the documentation of those others backplanes for instructions. On the module end, make sure that red band faces the end of the power header connecter closest to the marking RED BAND screened on the PCB, which in some cases - M41 included - will be facing downward.

FIGURE 3 • Power connection via flat cable

(A) Model 41 (note RED BAND marking facing down)

(B) Doepfer Backplane (note -12 marking facing down)

(C) 10/16 Flat Cable (note RED BAND facing down)



Before continuing once your flat cable is properly connected to your module, VERIFY THAT ALL SYSTEM POWER IS OFF. The second step is connecting the 16 pin end of the flat cable into one of the 16 pin power connectors on the Doepfer backplane, assuring that the RED BAND of the flat cable is FACING DOWN towards the -12 marking on the buss connector. Expect your Model 41 to draw 40 to 60 mA on the positive rail of your PSU. CUrrent to the neagtive rail is neglatiable, somewhere around 15mA. Your last step before powering up is securing the module in place by installing the four screws located on each corner using a Phillips head screwdriver. Pleas note that in some cases, your faceplate maybe become marred by the head of the screw. Plastic washers (not included) will stop this from happening. Once the flat connector is installed and your module is firmly mounted into your chassis rails, you're ready to run through basic and advanced features of the Model 41 Steiner 4P Filter.

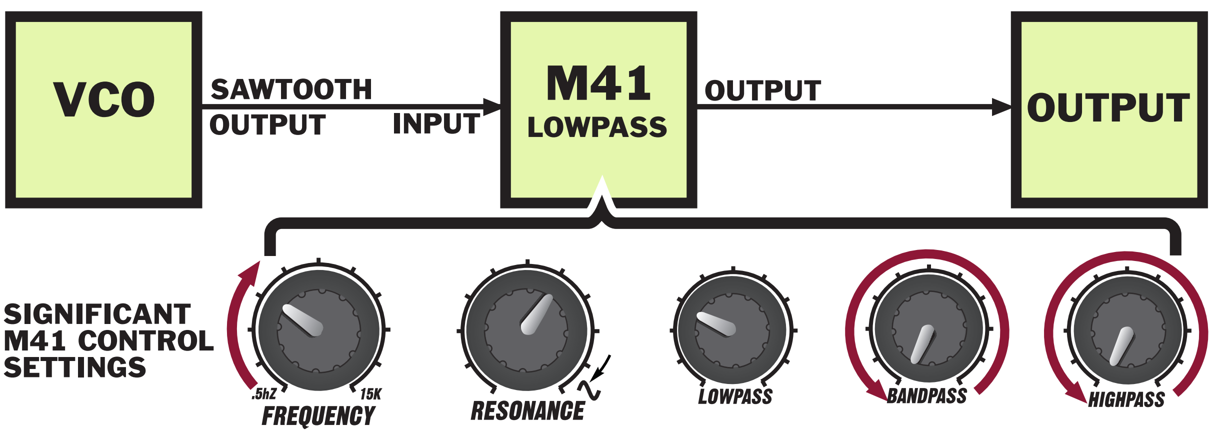
Basic Operation of the EAR Model 41

Like many instruments used in the generation of synthesized electronic sound, Filters can provide different basic functions: As **controllers** (audio band and LFO modulation source), as **sound sources** (sine oscillator) and what would be considered their primary function: **sound modifiers**. This chapter deals with the basic methods to incorporate the M41 to accomplish all three tasks.

Sound Modifier: The Model 41 Filter

The purpose of a sound modifier is to shape an AC signal to effect change to one or more of the five basic parameters of sound: frequency, amplitude, duration, timbre and placement (pan), with filters focusing on TIMBRE (pronounced TAM-BER) by subtracting harmonic content from the parent signal. Like all modifiers, the signal must be routed through a filter to effect change. The signal is inserted at THE INPUT JACK (page 6, # 14) and comes out at the OUTPUT JACK (page 6, #15). Once this is patched, set all manual controls to their null position (fully CCW) and all pushbuttons in the center of the module are disengaged (Leeds off).

FIGURE 9 • Basic Operation



The M41 has taken advantage of the unique Steiner core convention of assimilating it's three output modes by injecting the incoming signal into three different nodes of the circuit. This is accomplished by opening the input statement that correspond to those three modes: LOW, BAND and HIGH PASS. By doing this, it accords a forth NOTCH File mode by raising the level of the Low and High band ten to equal level and while figure uses the Always mode as an example, any of the three can be used for this test.

With the Frequency knob fully shut down, insert a sawtooth wave into the M41 and raise your Resonance Control to mid level to 3 o' clock setting to focus the resolution of the band sweep so that the individual harmonic partials of the input sawtooth wave begin to manifest. Raise the input attenuator of the selected band (LOW, BAND or HIGH PASS) to approximately 11 o' clock and then sweep the Frequency knob manually over it's full range of travel to hear the harmonic subtractive change of the three modes against the overtone structure of the parent signal. This is the basic of filtering.

Voltage Control

Like much of Plan B/EAR product catalog, the resting pulse is if there's a knob, stick a voltage control input on it and the M41 is no exception to this convention. Equipped with three individual VC inputs, both Resonance and Frequency of the Model 41 are fully voltage controllable, with the ladder having two inputs, each with it's own attneuator so that the user can take expect outcome results ranging from chaos to controlled, dramatic to sublimity subtle. There are two types of attneuators incorporated in the M41: standard unipolar 0 to 10 type and a second bipolar VC Input applied to the Frequency which allows for variable inversion of the incoming control signal, outlined below:

FIGURE 10 • Unipolar VS. Bipolar attenuation

A graphical representation of the resultant control effect of a single decay-only envelope upon:

A A UNIPOLAR control voltage input processor

B A BIPOLAR control voltage processor

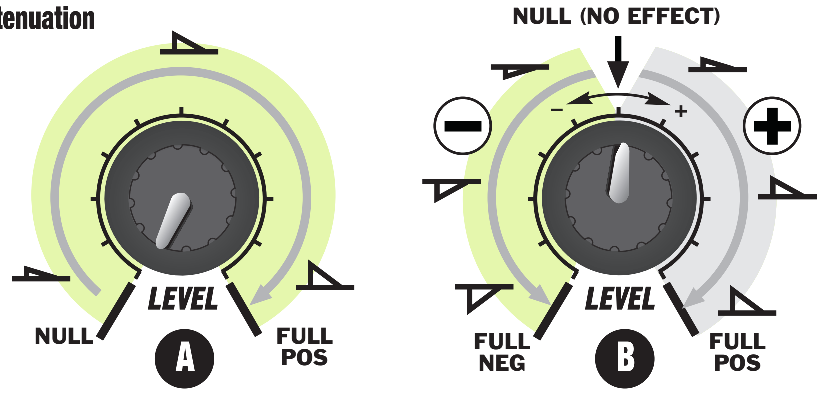
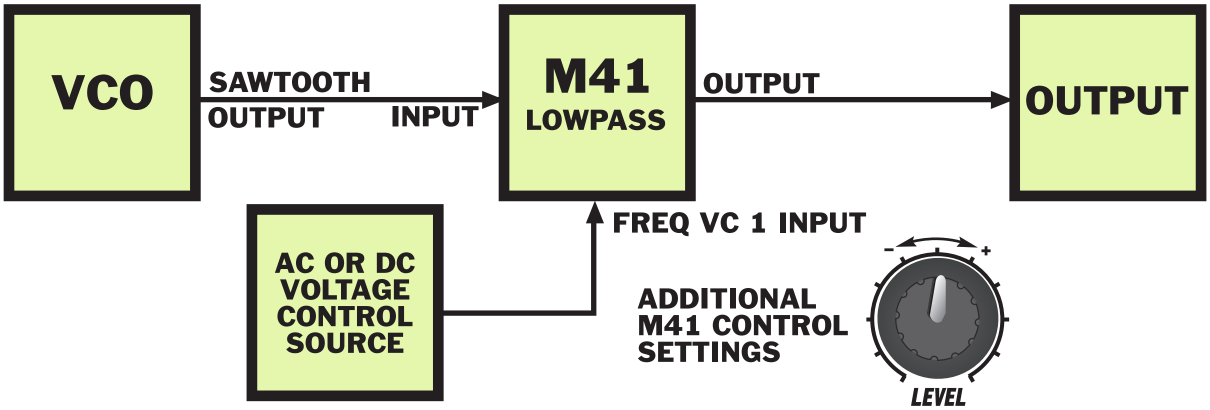


Figure 11 below outlines the basic configuration for applying a control voltage to effect automated change upon a given filter parameter which would otherwise be altered by varying the internal voltage manually. To do this, the user would insert a secondary voltage source, either AC or DC into one of three voltage control inputs on the M41: Freq VC1 (page 6, #3) Freq VC 2 (page 6, #5) or Res VC (page 6, #8) and determine the desired amount of change using the associated input attenuator control for each. The benefits of such an application are obvious: it allows for greater precision, greater speeds and increases the resultant timbral qualities of the parent sound being processed far beyond the power of mere mortals (wink).

FIGURE 10 • Voltage Control



Sound Source: The Model 41 Filter

The circuit configuration which allows Resonance (internally applied feedback) to a filter has a unique benefit in that a threshold will be reached when the filter core oscillates, generating some of the purest sine waves available in analog synthesis.

Sound Source: The Model 41 Filter (cont'd)

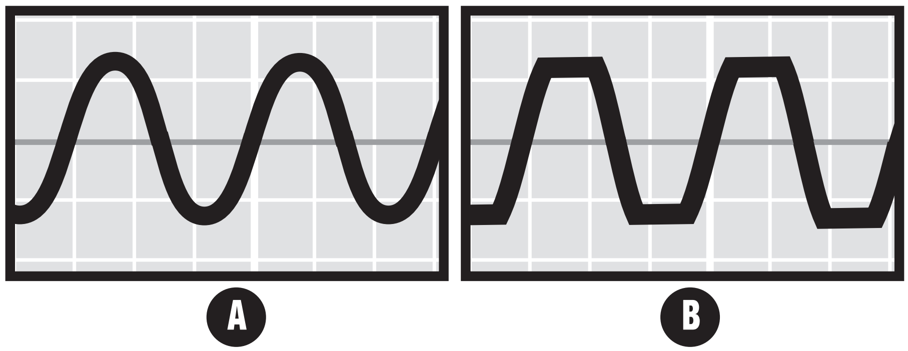
The Model 41 is calibrated so that oscillation occurs when the Resonance knob is set higher than 3 o'clock or when a control voltage forces the Resonance to exceed that limit with the manifest AC waveform can be determined as a pure sine wave to a clipped sine based on the amount the Resonance control setting exceeds that 3 o' clock threshold.

FIGURE 12 • Waveshape variations of the M41

The graphic indicates the resultant output waveform at 10 volts/div when:

A The Resonance is set at or near 3 o'clock

B The Resonance is set to max (fully CW rotation)



When the M41 is configured as a sound source, an input signal is not required - the user will simply patch the output to it's destination and turn up the Resonance until the desired output waveshape is achieved. However, do not disqualify inserting an input signal when is self oscillation because this is where some of the most complex timbres are achieved with the Steiner 4P Filter. 1v/octave tracking may be achieved by inserting a equal tempered keyboard controller in the FREQ VC2 control input of the M41 and adjusting the attenuator until correct scaling is achieved. Secondary frequency modulation can then be applied to the FREQ VC1 input without effecting the scaling set at FIRE VC2.

Controller: The Model 41 Filter (cont'd)

The oscilation range of the Model 41 is huge. On the high end, it will exceed the audio bandwidth to approx 16KhZ. On the low end however, it will range from between 4 and 5 octaves below the audio bandwidth limit of 16 hZ so that it can be used as a control LFO (Low Ferequency Oscillator).

Advanced Operation of the EAR Model 41

So far we have discussed the basic conventions of the Model 41 Steiner 4P Filter, which apart from it's own sonic charactor does not suffiecient isolate it from many other filters on the market.   However, there are three unique functions on the M41 that have never been oncorporated into any filter, which dramatically increase it's timbral footprint: IFM Sync, Hyper Q and a variatible Sprectral Energy Distribution manual control.

Along the lower center section of the M41 you will find a series of four latched and illuminated pushbuttons. This is where the critical features of the M41 IFM Sync (page 6, # 9) Hyper Q (page 6, #12 and 13) and Spectral Energy Distribution (page 6, #10) are engaged. Two of these (IFM Sync and Hyper Q) are internally patched control configurations which induce two unique distortions modes, one applied against the frequency and one the resoanace.

Advanced Operation of the EAR Model 41 (cont'd)

IFM SYNC
To begin testing the effect of the IMF Sync, configure the pushbuttons so that none are illuminated, and then engage the IFM SYNC mode be depressing the top pushbutton so that it is illuminated. When this is done, an internal patch has been engaged to the frequency control circuit which manifests quantized harmonic resolution above and beyond when can be achieved with the Resonance pot and introduces a SECONDARY RESONANT FREQUENCY harmonically constituent to the the first - dramatically effecting the output timbre which, when offset by the Resonance pot setting, ranges between pretty to bad-hair-day Godzilla.

HYPER Q
The Hyper Q distortion mode is similar to IFM Sync, but applied to the Resonance control circuit. Because of the complexity involved with this internal patch two pushbutton variables are required. To engage Hyper Q, depress the third pushbutton (page 6, #12) until it illuminates while the other three are off. Insert a sawtooth wave into the M41 and monitor it's output. This is where the second polarity pushbutton is addressed: If your sawtooth is positive going - the Polarity pushbutton (page 6, #13) should NOT be engaged. If your sawtoth is negative, it should. If you are not sure of the phase of your parent waveform- try both settings of the Polarity pushbutton and it will become immediately obvious which is required.

Combined IFM SYNC and HYPER Q

No explanation nessesary...just try it (wink).

SPECTRAL ENERGY DISTRIBUTION
The mark of a filter is determined not only by it's ability to rip an input parent signal to shreads, but also it's ability to induce subtle and natrual timbre modifications. Anticipating that, the M41 has been equipped with a variable Spectral Energy Distribution control which lessens the intensity of the filtering without the phase problems associated with a standard wet/dry crossfader. SED does not illiminate filtering - it affords the user a non-arbitrary means to decrease and increase the depth of the effects of filtering from full to rubbery and soft. When engaged, a certain amount of the low end audio bandwidth will attenuate approx. 10dB so an enabler has been included to take the control in and out of the circuit altogether. To engage, depress the ENABLE pushbutton and set your desired SED level manually with the it's pot to the desired level. When this button is not lit, the SED is out of circuit and will have no effect. Make an effort to realize that subtle is not bad. It's the difference between pastel and acrylic paintings - both having their place in the art world equally.

VIRTUAL BAND-REJECT/NOTCH FILTER
By combining the Lowpass and Highpass attenuators to an equal audio level while keeping the Bandpass level to fully CCW (null), you can effectively manifest a forth output mode: Band-Reject or Notch Filter.

Warranty Information • Model 41

The Electro-Acoustic Research Model 41 Steiner 4P Filter is being built and distributed through a arrangement with Darkplace Manufacturing - a subidiary of Malekko Heavy Industry - in Portland, OR., and as such is warrantied for one year after the date of purchase for all defects due to manufacturing error. Repairs should be sent to:

Darkplace Manufacturing
814 SE 14TH AVE. PORTLAND OR 97214
+1-503-406-2770

Calibration Procedure • Model 41

The Model 41 undergoes extensive set-up to tune each module to it's highest possible performance standards. Therefor, changing these settings is not recommended. Instead, apply your curiosity to more productive crusades - like designing amazing sound with the amazing tools now at your disposal.

1) Set the CV Reject Trimmer to mid point and set all pushbuttons to 'off' (LED unlit)
2) Set the Res Bias Trimmer and Max Res Trimmer fully CW.
3) Turn frontpanel Freq pot fully CCW
4) Set frontpanel Res pot fully CCW
5) Set HIgh VC Trim Trimmer fully CCW
6) Freq Low VC tirm Trimmer fully CW
7) Insert a sawtooth wave into the filter with a frequency of 20 Hertz and an output level into the filter of 10 volts peak to peak (Doepfer std amplitude level)
8) Bring up the Lowpass input attenuator to 11 o'clock. Bring the Bandpass and HIghpass attenuators fully CCW
9) Turn frontpanel Freq knob to 9 o' clock
10) Turn the VC Low Trim trimmer until an output is heard, then back it off until the signal is not longer audible
11) Turn the front panel Freq pot fully CW
12) Turn the VC High Trim trimmer until the filter begins to audibly change frequency
13) Turn frontpanel Freq knob to 2 o' clock
14) Turn frontpanel Res knob fully CW
15) Turn Res Bias Trimmer fully CCW
16) Remove the input signal from the filter
17) Set the Max Res Trimmer until the filter begins to audibly go into oscillation

Calibration Procedure • Model 41 (cont'd)

18) Set the fontpanel Freq knob to 12 o' clock

19) Turn the Max Res trimmer until a full frequency range self oscillation is achieved

20) Using an Oscilloscope, verify that the oscillation ranges from approx .5 hz to approx 15KhZ

21) Set the frontpanel Res pot to 2 o'clock

22) Turn the Res Bias Trimmer until the self oscillation stops

23) Reinsert the 20 hertz input signal

24) Fine adjust the Low and High Freq trimmers until the max range of the frontpanel Freq pot is achieved with and the has an audible range with the output first opening when the the Freq pot is set to the first tick mark above fully CW and fully opens then the frontpanel Freq pot set to the last tick mark before fully CW.