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User's Manual

Version 0.50 of User's Manual for AFM-1 version 1.01

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TABLE OF CONTENTS

(TABLE OF CONTENTS, CONTINUED)

Distortion and resonance have a history that reaches back to the origins of language and music, finding an antecedent in overtone singing which is thought to have originated in south western Mongolia. This makes resonant filters a prime example of convergent musical evolution. Given the time scales involved, it's worth considering why these sounds are so evocative. Distortion is a non-linearity that changes the timbre of a sound in response to its volume. The human voice, and many other instruments, cannot easily change volume without also changing timbre. This is also true of analogue filters. The same may also be said of the electric guitar: Early amplifiers, regardless their quality, would distort with increasing volume, fulfilling the metaphor of the human voice at its most primeval. Articulated resonance is a relatively recent development, from a time after voices were first heard, but – perhaps – before language. These ingredients are found in music-making, performance, and, now, in electronic circuits that can be constructed and modified with relative ease. The more accessible these materials become, and adaptive to the fascination of curious minds, the more they give voice to previously unheard aspects of thought, just as language itself endlessly evolves to improve our collective capacity for understanding.

INTRODUCTION

This manual describes the AFM-1, a reference design for an audio filter circuit. The design comprises a schematic and this manual. Recommendations are provided for component values, and the types of components used, for example metal film resistors, polystyrene capacitors, op-amps, and so on. With this combination of components, the circuit has a characteristic sound. This reference design is intended as a set of instructions for obtaining this sound, although many modifications to the circuit are possible, and encouraged.

For example, the use of LM1458 op-amps is recommended for no other reason than the way that they distort. From a traditional design perspective, these op-amps are out-dated, consume unnecessary amounts of power (which may be an issue for battery-operation), and add distortion and noise that could easily be avoided using a compatible alternative. However, the AFM-1 is *intended* to distort, and to do so in a way which is expressive.

Simply choosing components that distort is not enough: It is necessary for the distortion to be appropriate. For example, polystyrene capacitors have been chosen for two of the most critical components in the audio path, because these capacitors distort *less* than ceramic capacitors, whereas the output stage of the circuit includes diodes configured to generate as much distortion as possible. There are a number of electrolytic capacitors in the audio path, because these *do* have non-linearities that have some value in this circuit. Furthermore, electrolytic capacitors degrade over time, and so this circuit embraces the idea of decay, as well as distortion.

Other component characteristics which affect the sound include the noise generated by metal film resistors, which is less than that of carbon construction resistors, but of a different character. Different resistor materials give noise with different properties, such as the relative probability of high magnitude flickering. The noise characteristic can affect the way the state of the filter evolves over time, nudging it into a particular sequence of behaviours according to a momentary magnitude. Carefully crafted noise is an essential ingredient of filter characteristics.

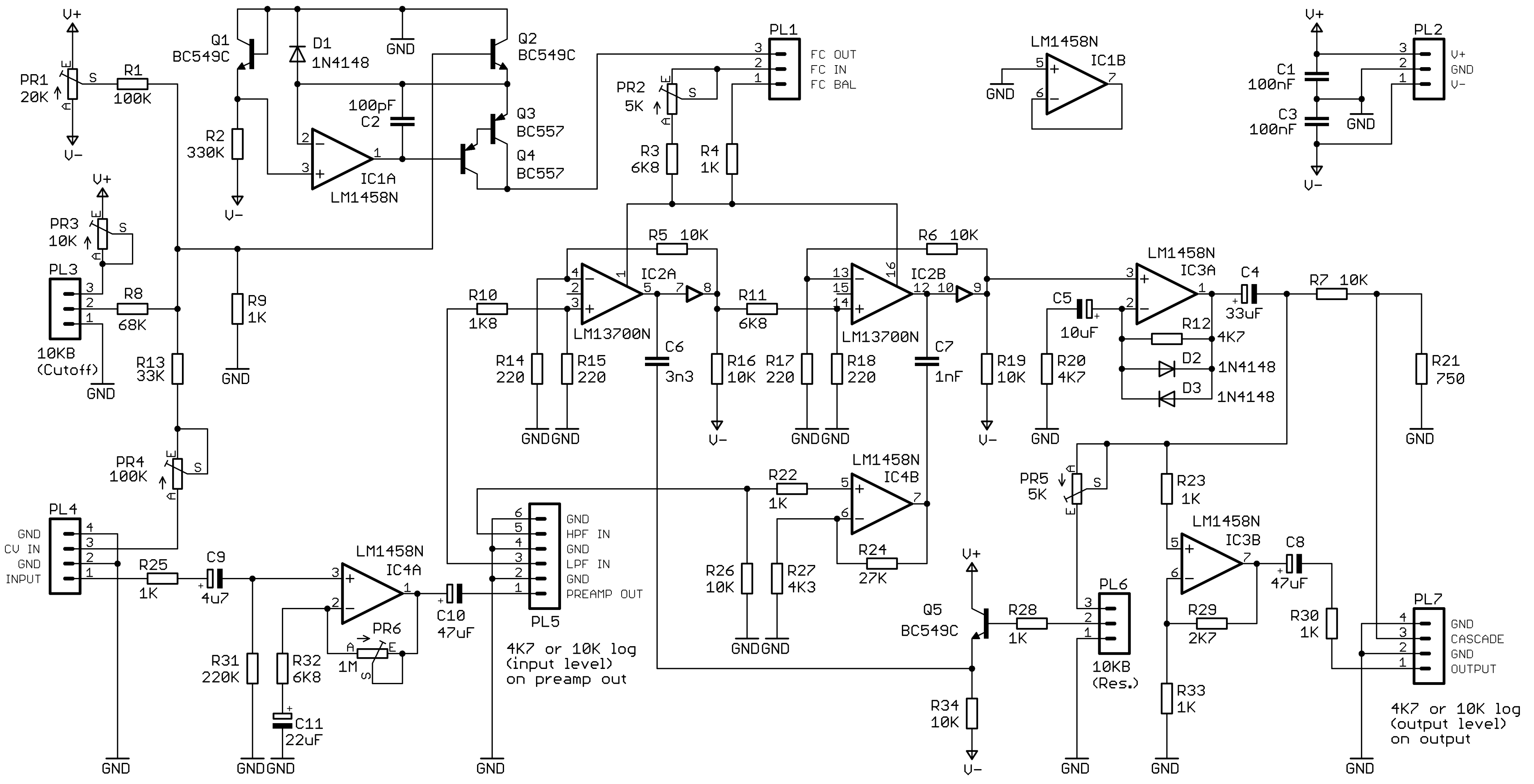
The AFM-1 provides a common point of reference, specifying components for a very particular set of sound characteristics. It provides a starting point for many different variations, hopefully including those which are easier to construct, as well as those that are much more complex. The design files, for the schematic and the circuit board layout, are available as computer software under the GNU General Public License (GPL), and this manual is provided under the terms of the GNU Free Documentation License, with the aim of encouraging further investigation.

BACKGROUND

The AFM-1 is inspired by the 12dB-per-octave filter used in the Korg MS10 and MS20 synthesizers, manufactured in the 1970s. The MS10 was a single low pass design, the MS20 had a high pass filter followed by a low pass. The circuit used in these instruments was revised, so there were two different types of filter used, with slightly different characteristics. It's generally believed that the first version is better.

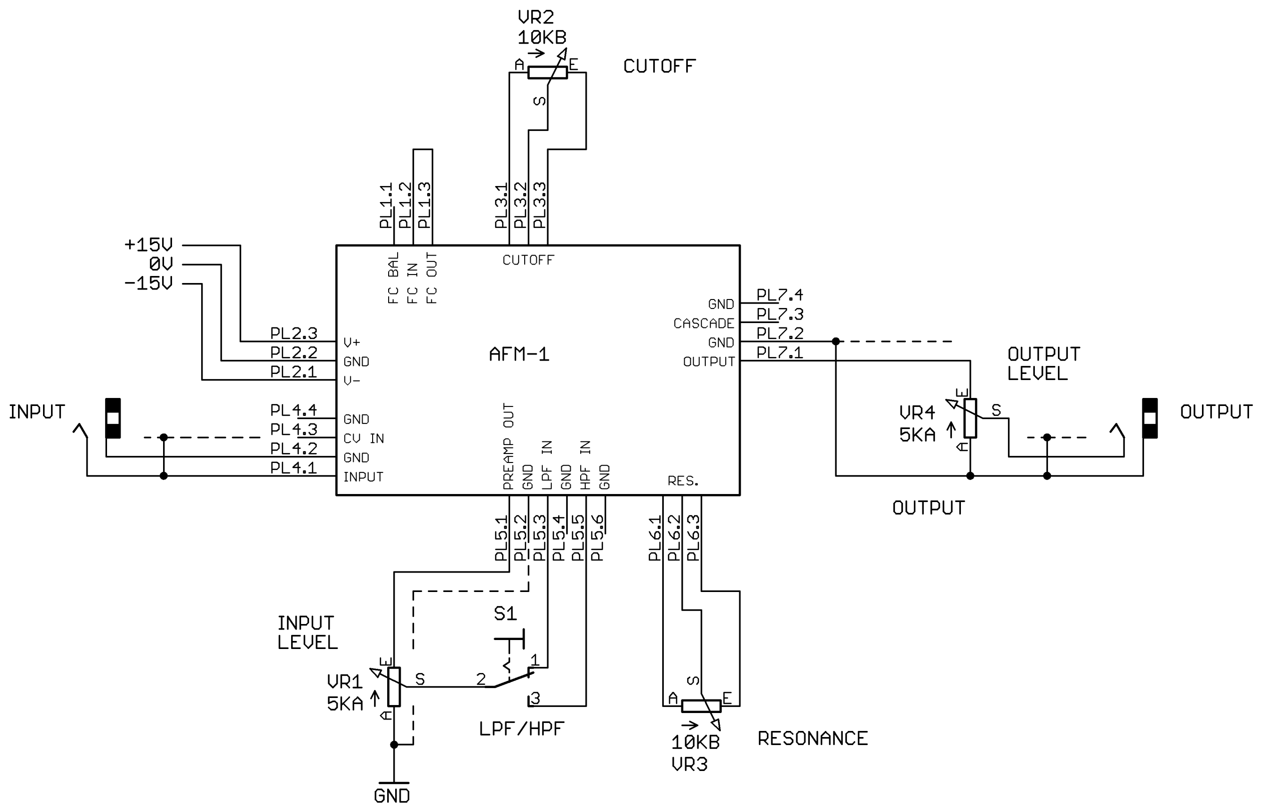
In 1997, I made an attempt to transcribe the first version of the circuit from an MS10. This included a custom Korg-35 IC, whose contents, at the time, were unknown. Comparing this version of the circuit with the later version in another MS10, various incorrect assumptions about the Korg-35's topology were made. The resulting circuit sounded similar to both, while at the same time having an unusual character of its own. In particular, the two cascaded stages of the AFM-1 have different characteristic frequencies. This has turned out not to be a feature of either version of the Korg filter. The circuit was then modified to distort more readily, by arranging for self-oscillation to occur at a level closer to that of the signals being filtered.

Before and since the AFM-1 was designed, many MS10/20-inspired filter circuits have been produced in the form of synthesizer modules. Some of these are extremely accurate reproductions of the original Korg-35-based circuit, while others are based on the later version. More recently Korg has started manufacturing analogue products based on the MS10/20, fully recreating the Korg-35 filter circuit using discrete components.



Analogue Filter Module

TITLE: AFM-1	
Document Number:	REV: 1.01
Date: 5/06/2006 19:22:50	Sheet: 1/1



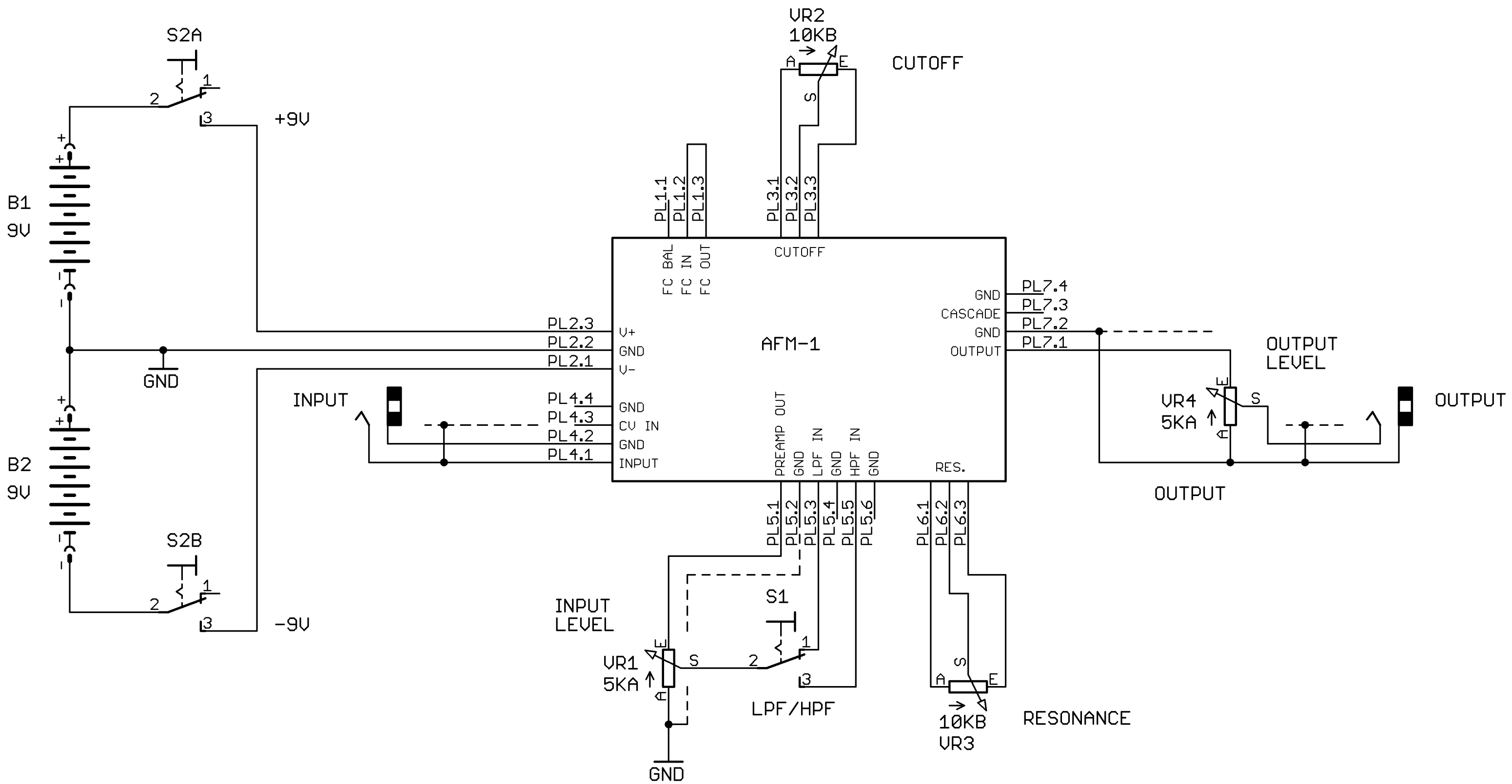
Power requirements: 25mA at +/-15V

Pots marked with 'A' are logarithmic (eg 5KA)

'B' indicates a linear pot (eg 10KB)

Connections for single AFM-1 module

TITLE: Single Module	
Document Number:	REV:
Date: 5/07/2006 22:28:52	Sheet: 1/1



Power requirements: 10-12mA at +/-9V, 7mA at +/-7V (partially discharged)

PP3/6LR61 alkaline batteries: around 50 hours' continuous use

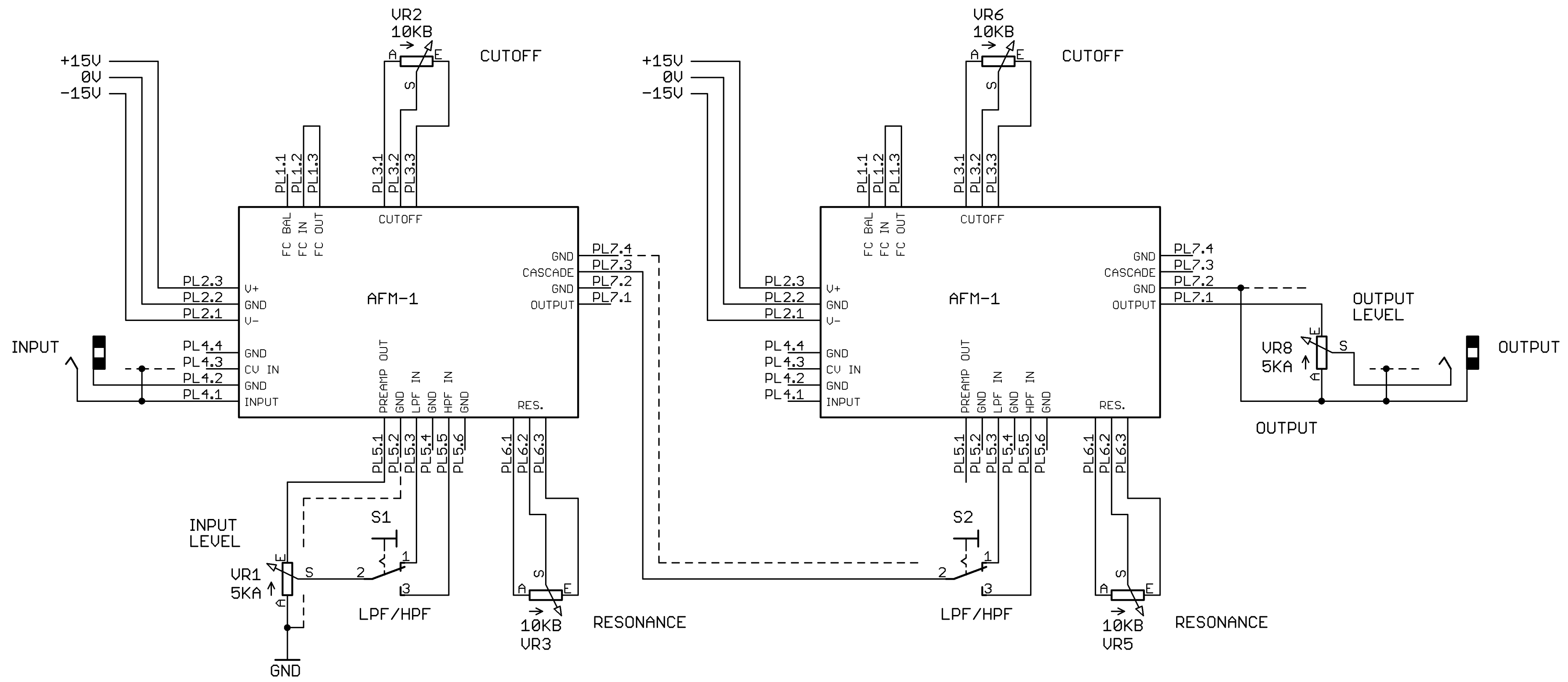
PP3/6LR61 170 mAH NiMH rechargable batteries: around 17 hours' continuous use

Low voltage AFM-1 cutoff range is improved by changing R8 to 47K

(for example, by placing 150K in parallel with existing 68K)

'A' indicates a logarithmic pot (eg 5KA), 'B' indicates a linear pot (eg 10KB)

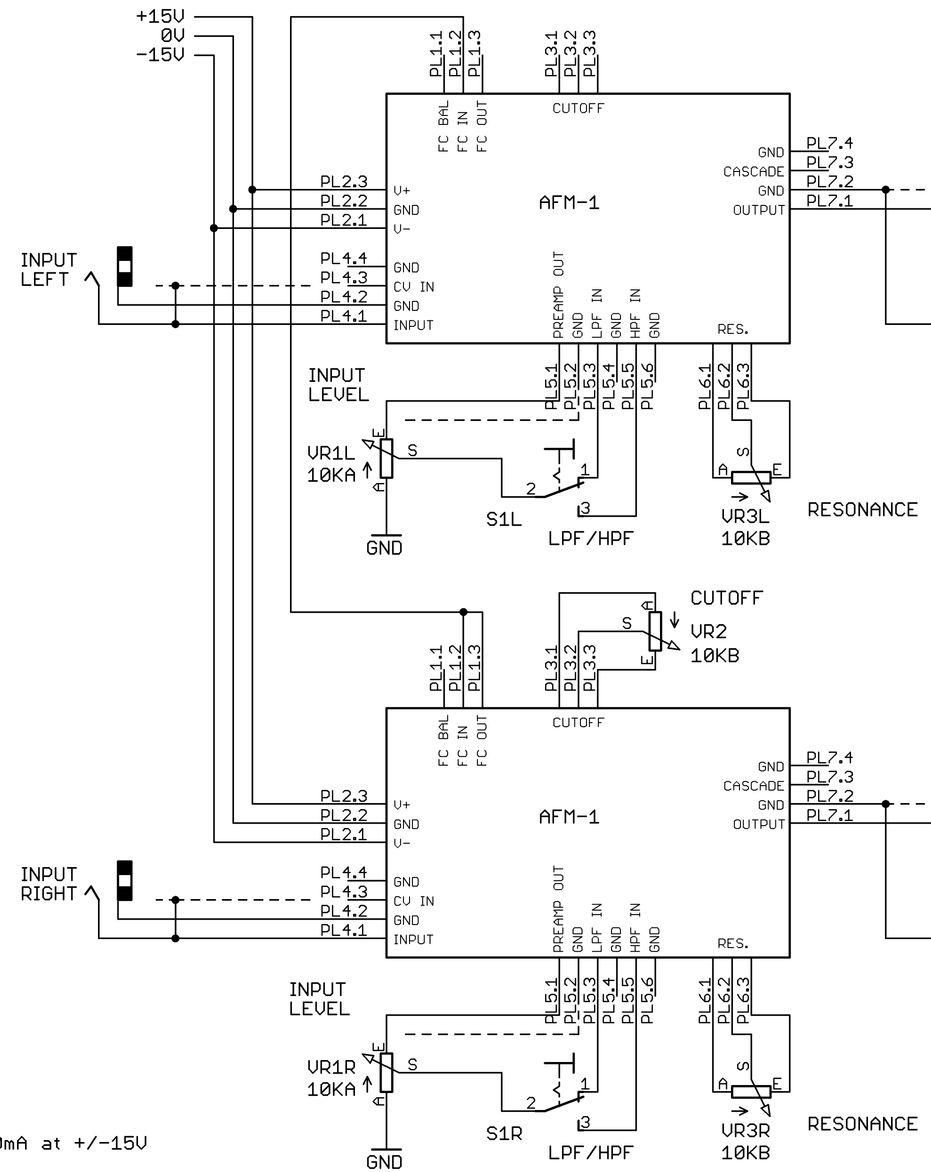
Battery power for single AFM-1 module	
TITLE: Single Module Battery Supply	
Document Number:	REV:
Date: 7/26/2006 19:43:20	Sheet: 1/1



Power requirements: 50mA at +/-15V

Pots marked with 'A' are logarithmic (eg 5KA)
'B' indicates a linear pot (eg 10KB)

Connections for cascaded AFM-1 modules		
TITLE: Cascaded Modules		
Document Number:		REV:
Date: 5/08/2006 20:14:18		Sheet: 1/1



VR1L/R, VR3L/R and VR4L/R are dual-ganged stereo pots.

10KA values are used for VR1 and VR4 as dual 5KA are hard to obtain.

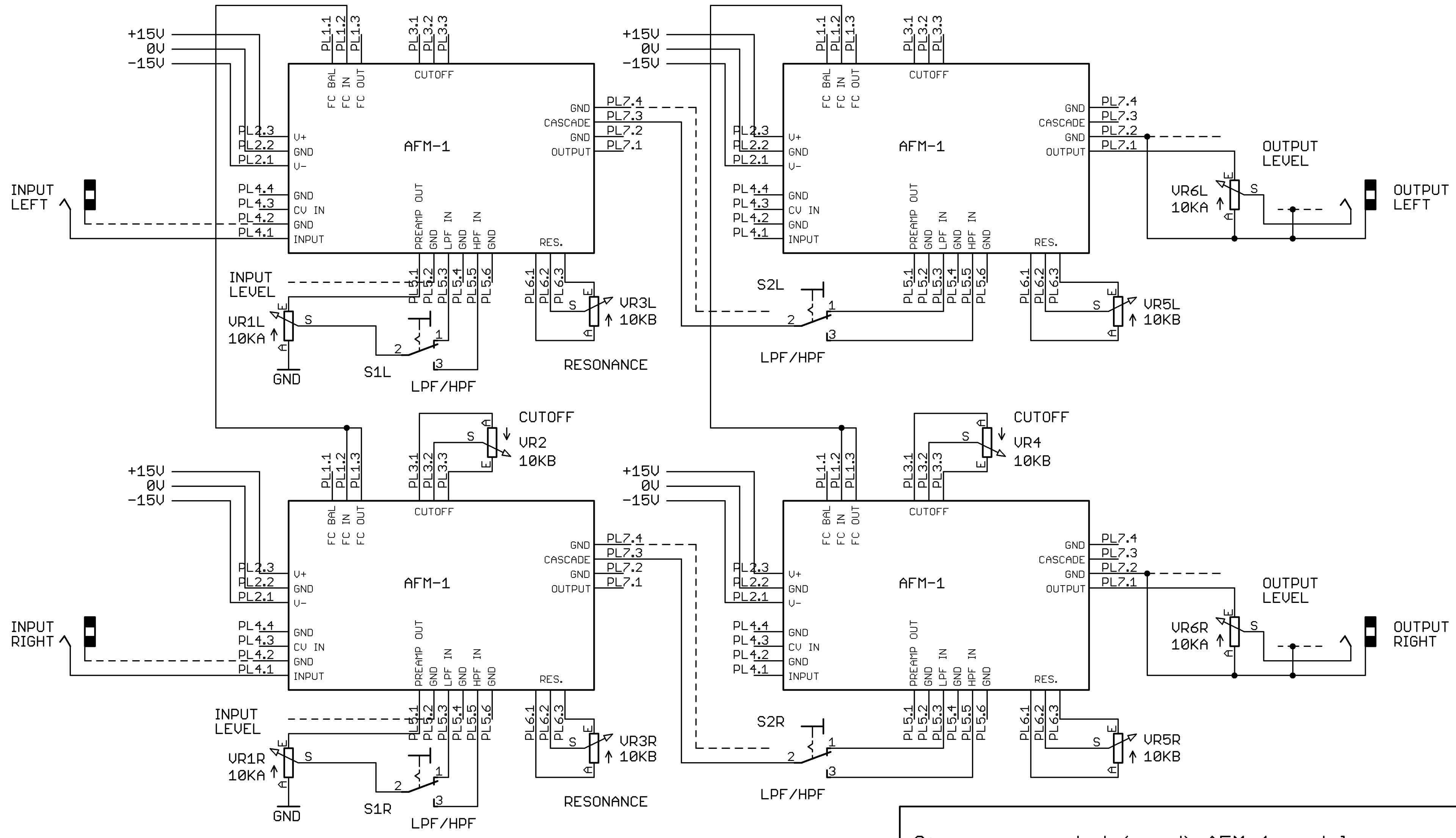
Frequency control is derived from VR2 and the resulting frequency control signal (FC OUT) is supplied to the other module.

See additional notes for cutoff balance control.

Power requirements: 50mA at +/-15V

Pots marked with 'A' are logarithmic (eg 10KA)
'B' indicates a linear pot (eg 10KB)

Connections for stereo AFM-1 module pair	
TITLE: Stereo Dual Modules	
Document Number:	REV:
Date: 5/07/2006 22:28:24	Sheet: 1/1



Power requirements: 100mA at +/-15V

Pots marked with 'A' are logarithmic (eg 10KA)
'B' indicates a linear pot (eg 10KB)

VRnL/R are dual-ganged stereo pots

10KA values are used for VR1 and VR6
as dual 5KA are hard to obtain.

See additional notes for cutoff balance control.

Stereo cascaded (quad) AFM-1 modules	
TITLE: Stereo Cascaded Modules	
Document Number:	REV:
Date: 5/07/2006 23:21:41	Sheet: 1/1

SPECIFICATIONS

Power Supply

±15V at 20-25mA per board (recommended)
±12V at 17-20mA per board
±9V at 10-15mA per board (see Battery Operation, page 16)
±7V at 7mA per board (for example, partially discharged 9V batteries)

Input Preamplifier

Optional preamplification stage contained in the filter module, typically used in the first of a cascaded chain of filter modules.

Input signal (recommended): 50mV to 1V RMS (adjustable gain)
Input impedance: 220K ohms

Filter

12dB-per-octave low pass filter (LPF) or 6dB-per-octave high pass filter (HPF).
Filter exhibits resonance, self oscillation, distortion and phase-locking.
Input signal range (recommended): 200mV-4V RMS.
Input impedance: 2K ohms (LPF), 10K ohms (HPF).

Exponential Converter

Linear (pitch) to exponential (frequency) converter, cascadable to other modules for parallel tracking, with pitch offset (frequency balance) capability.

Cutoff potentiometer input (0V to 15V).
Control voltage (CV) input (adjustable sensitivity)

Output

Level: 2V-3V RMS (at self oscillation)
Impedance: 1K ohm
(Capable of directly driving headphones having impedance equal to or greater than 32Ω)

APPLICATION NOTES

The AFM-1 has been optimised for distortion, as this is the enduring strength of circuits of this kind. A single filter is perfectly capable of radically shaping an audio signal on its own. However, the AFM-1 was designed to be used in a variety of configurations, switching easily between high pass and low pass characteristics, and easily supporting stereo and cascaded configurations, as shown in the preceding schematics.

Alternative Module Configurations

The module configuration schematics are intended only as a guide to using the filter module. In many cases a minor variation in the configuration will provide something useful. For example, in the Stereo Cascaded (quad) AFM-1 example, a DPDT bypass switch can be provided to exclude the second pair of modules from the signal chain. In this case, the switch would route the top terminal of VR6 from the PL7.1 OUTPUT connection of the first or second module in each of the chains.

When cascading modules, the cascade output (from PL7.3 CASCADE) doesn't have to be used. It is also possible to use th PL7.1 OUTPUT connection and supply this to PL5.3 LPF IN or PL5.5 HPF IN via a potentiometer. This makes it possible to control the amount of distortion generated in the cascaded module independently of the distortion of the preceeding module in the chain.

Multiple independently configurable modules may be used, in which the possibility of linking and routing for stereo (or other parallel) operation can be provided by switching. It may also be useful to be able to bypass or include filter modules in the signal path using various switching options. Switching can be performed using toggle switches, switches on push-pull potentiometers, jack socket switches, and so on.

Using the Preamplifier

The preamplifier, based around IC4A, provides an adjustable level of gain, and is required to bring the level of incoming signals up to that required for best filter operation.

The preamplifier output (PREAMP OUT) is provided at pin 1 of PL5. At this point in the circuit, a logarithmic potentiometer can be used to adjust the filter input level. The wiper of the potentiometer is supplied to either pin 3 (LPF IN) or pin 5 (HPF IN), typically via a switch.

If the input level potentiometer is placed at the input of the preamplifier, and a direct connection made from PREAMP OUT to LPF IN or HPF IN, noise from the preamplifier will be conspicuous at low input levels.

The filter output, provided at pin 1 of PL7, is a high gain output, usually supplied to an output level potentiometer. By modifying the input level and compensating with the output level, the amount of distortion generated by the filter can be varied widely. The output, even via a 10K potentiometer, can drive headphones directly, and this may be useful for monitoring the circuit.

When filters are cascaded, the preamplifier and level potentiometer of cascaded stages can be omitted. In this case, the output should be taken from pin 3 (CASCADE) of PL7, where the level is set so that cascaded filters receive about the same level of signal, and distort to roughly the same degree. In this case, the CASCADE output is connected directly to LPF IN or HPF IN.

If different levels of distortion are required for each filter in a chain, then use pin 1 (OUTPUT) of PL7, and a logarithmic potentiometer to set the amount of level received by LPF IN or HPF IN on the next module.

Adding a Cutoff Balance Control

The examples shown in the schematics exclude a cutoff balance control. This can be used to adjust the relative cutoff frequency of a pair of modules that share the same exponential converter (i.e., the FC OUT from pin 3 of PL1 is supplied to FC IN on the same module, and also to FC IN on a second module).

Use a 5K ohm linear potentiometer. Connect the wiper to FC OUT, one side to FC IN and FC BAL and on the same module, and the other side to FC IN and FC BAL on the second module. The centre point of the potentiometer can be trimmed for equal cutoff using PR2 on one or both the modules. Different potentiometers give a different balance range.

Battery Operation

The AFM-1 can be powered from a dual 9V supply provided by two batteries, as shown in the schematic on page 10. In this case a power switch is required for both the positive and negative supply lines. A DPST switch is suitable. Alternatively, the positive side of the supply can be switched on the input jack socket, and the negative side of the supply can be switched on the output jack socket. Power consumption at ±9V is around 10-12mA per rail, and less than this after the batteries have discharged somewhat. A pair of alkaline PP3s will give around fifty hours' use. A pair of fully charged 170mAH NiMH batteries should give more than 17 hours' continuous use. Do not solder direct to the batteries.

The circuit works normally at the lower supply voltage, although the op-amps will distort more easily. The op-amps have been chosen for their distortion characteristics, so the lower voltage may lead to an preferred distortion aesthetics, for which it may even be worth running off a reduced supply voltage even when a higher voltage is available. A higher supply voltage concentrates distortion in the diodes D2 and D3, whereas a lower voltage increases the proprtion of distortion contributed by op-amps. Distortion that arises in many parts of the circuit is very different from an increased level of distortion from just one part of the circuit, for which the circuit is primarily designed.

However, there is a slight disadvantage when operating from a dual 9V supply, and this due to the cutoff circuit having less range. As a result, the highest calibratable cutoff frequency is reduced, although this may not be noticable. A simple solution is to change R8 to 47K. This is easily achieved by soldering a 150K resistor across the existing R8 68K resistor on the reverse side of the circuit board.

It is tempting to provide a power indication LED, but this can use up a lot of current. If an LED is required, use a 2mA low current type, in series with an 8K2 resistor across the full 18V of the ±9V supply . This will reduce battery life by around 20%.

Single Battery or External Power Supply

It is inconvenient to provide two batteries, and furthermore makes it impossible to use an ordinary external 9V power supply adaptor. One solution to this is to use a power supply inverter circuit. A circuit of this kind can be constructed around an inverter chip such as the ICL7660S, set to a high internal oscillation frequency to avoid audio interference. This circuit can generate a -9V supply from +9V, and operates with high efficiency. Keep in mind that the current requirement from the single supply is then doubled, to 20 to 25mA, so when operating an inverter-equipped circuit from a single battery, battery life will be half that of when two batteries are used. For example, a single alkaline battery would give around 25 hours of continuous use, and a fully charged NiMH battery will provide somewhat less than this.

CIRCUIT DESCRIPTION

Preamplifier

The preamplifier, based around IC4A, provides gain for input signals. Its output is usually coupled to HPF IN or LPF IN by connections made to PL5. An LM1458 op-amp has been used, because of its preferred distortion characteristics. The LM1458 is effectively a dual 741. More up-to-date op-amps are either somewhat transparent, or impart very unpleasant distortion when a particular output level is reached. A 741 always distorts slightly (a few fractions of a percent THD at least), and this distortion builds up smoothly as the chip tries to reach its maximum output. A 741 is also noisy, so the input level should be varied using a pot on the output of the preamp, rather than the input. The noise can be valuable for disturbing the stability of the filter at high levels of resonance, so it would be incorrect to assume it has no positive contribution to make. The LM1458 has a particular set of sonic characteristics, which, although not ideal in the usual sense, make a positive contribution to the sound of this circuit. Different pin-compatible op-amps can be tried out in the position of IC4, but this is not particularly interesting – recommended experiments are described later.

Low Pass Filter

The filter is based around the two operational transconductance amplifiers in IC2. It is easiest to consider the low-pass configuration first. The filter is effectively two buffered 6db-per-octave RC sections. In the first section, IC2A forms the resistor, whose value is controlled by the current into pin 1, and C6 forms the capacitor. C6 is effectively pulled to ground by the emitter of Q5. A high frequency signal sees C6 as having a lower impedance than low frequency signals, and so the junction of the resistor and the capacitor, at pin 5 of IC2A, has a 6dB-per-octave low pass characteristic. A buffer between pins 7 and 8 ensures that this characteristic is not smoothed-out by the next section. In the second filter section, the variable resistor is IC2B, set to the same value as IC2A, but the capacitor, C7, is one third the size of C6, meaning that the low pass characteristic starts at a higher frequency, but still having a 6dB-per-octave slope. C7 is also effectively connected to ground, in this case at the output of IC4. So these two low pass sections combine to give a 12dB-per-octave slope. Because the two sections have a different starting frequency, the slope characteristics are complex.

In particular, the phase (or time lag) of different frequencies varies in a more complex way than if the two filter sections were identical. Even slight variations – resulting from the expected component variations in manufacture – can cause significant variations of this kind. However, in this case, the two sections are very different, due to the 3:1 ratio of C6 to C7. When the filter is at high levels of resonance this provides an increased level of complexity.

The buffered output of the two filter sections is supplied to an amplifier stage, based around IC3A, having a gain of 2. However, this gain is limited by a pair of back-to-back diodes, D2 and D3, that have the effect of limiting the output voltage swing of this amplifier. Using analogue terminology, the diodes are said to perform “clipping”. However, this term needs to be used with caution when comparing with signals in the digital domain, which may also be clipped. The characteristics are very different: Digital clipping presents a discontinuity, a specific, abrupt numerical value, above which the output cannot increase. Analogue clipping is a loose term used to describe the exponential conduction characteristics of a semiconductor PN junction. In practice, this results in a curved transfer function, that smoothly changes gradient as conduction increases in response to voltage applied across the junction. A further difference with the digital domain is the regular sampling rate, that causes additional artefacts to be created that would not be present at all in the output of an analogue distortion circuit such as this.

The distorting amplifier based around IC3A is nothing special until resonance comes into play. The output from IC3A is supplied, via a resonance potentiometer and a buffer (Q5) to the other side of C6. Feedback at low frequencies is low, because C6 conducts high frequencies better than low ones. Feedback at high frequencies is prevented by the low pass action of IC2B in conjunction with C7. However, there is a frequency between these two extremes where positive feedback can be greater than unity, and also where the phase lag for that frequency is around ninety degrees. This effectively configures the filter as a sinusoidal oscillator, where the forward path is ninety degrees out of phase with the feedback, so the two push each other round indefinitely. With a gain greater than one, the amplitude would increase indefinitely were it not for the limiting effect of the diodes around IC3A, distorting and flattening the peaks of the sine wave. If the gain set by the resonance potentiometer is increased further, the amount of flattening of sine peaks is increased, resulting in audible distortion of the sine wave. This is assuming that there is no input signal, and the oscillation has been triggered by noise generated in the pre-amp or elsewhere in the circuit.

When input signals are combined with high levels of resonance at or beyond self-oscillation, the filter can act as a phase-locked loop. However, the complexity of this circuit, the potential number of sources of non-clipping distortion (the countless transistors in the op-amps), means that phase locking is only the most easily identified distortion that this circuit (and other analogue resonant filters) can produce. The significant levels of noise in this circuit make phase-locking and de-locking smoother and more random than would otherwise be the case.

High Pass Filter

Signals supplied to HPF IN (pin 5 of PL5) are amplified by IC4B and supplied to the other side of C7. From the point-of-view of the output of IC4B, C7 forms a high pass RC filter with the variable resistance of IC2B. Signals supplied in this way, to C7, end up having a 6dB-per-octave high pass filter characteristic at the output of the filter. 6dB-per-octave may seem a little disappointing, but resonance adds a lot to this characteristic, giving it all the complexity of the low pass filter that has already been described.

Optimisation for Distortion

Although many parts of the circuit add their own subtle distortions, the main source of distortion is the diode clipping circuit based around IC3A. What is particularly worth avoiding, is clipping distortion that occurs outside the feedback loop of the filter. For example, if the preamplifier was driven to distortion, then this would not contribute to filter complexity: Such distortion wouldn't be part of the filter function. Furthermore, given that the smooth clipping performed by diodes D2 and D3 is considered the primary source of distortion, it would be unfortunate to have earlier circuits distort significantly, limiting the signal, and thereby limiting the difference that D2 and D3 can make. In practice, this requires careful definition of the gain at different parts of the circuit, to ensure that the highest possible unclipped signal levels are supplied to the final clipping circuit containing D2 and D3.

A first distortion optimisation is at the input to IC2A. In most filter circuits that use the LM13700 or LM13600, the input resistor R10 is 10K. In conjunction with R15 at 220 ohms, this divides the signal level by a factor equal to the gain set by R5. The purpose of this arrangement is to reduce the signal to levels where the operational transconductance amplifier (OTA) can operate in its most linear region, but the overall gain, at the output of the OTA, is unity. In the AFM-1, R10 is set to 1K8, giving the first filter section a gain of around 5. This gain does not have to be used: correspondingly lower signals can be supplied to the filter. However, the alternative is to set the gain of the pre-amplifier five times higher. One can overlook the fact that a 741-style op-amp may not have a fast-enough slew rate to handle the additional gain: The problem is that the power supply is limited, resulting in clipping distortion, before the signal gets to the filter. With the first filter section having a gain of 5, it is certainly possible that it may distort, but at least this occurs in circuitry that can interfere with filter feedback, thereby increasing the complexity of filter characteristics.

The second filter section also has some gain, set by R11 to around 1.4. This is necessary to increase the gain of the resonant feedback loop, so that high resonance is easily attainable. It is also advantageous because it is providing gain later in the signal processing chain, pushing back the point in the circuit at which the power supply starts to limit signal amplitude. If such limiting does occur, it will probably occur within the feedback loop (assuming some degree of resonance), and if these limitations don't occur, as is intended by this design, diodes D2 and D3 will be entirely responsible for any large scale clipping that does occur.

The filter therefore has an overall gain of 5 x 1.4 = 7. The high pass input, at C7, must match the level of low pass signal that is expected at the filter output, otherwise switching between high pass and low pass will result in a level mismatch, and the two configurations will have different distortion characteristics, as well as levels. To balance the high pass and low pass output levels, the gain of the filter, 7, is approximately matched by the gain of the amplifier formed by IC4B.

Finally, the limited, distorted output from IC3A is amplified by a factor of 3.7 by IC3B. The purpose of this is to ensure a wide range of distortion characteristics are possible. With low input signals, the output of the filter needs to be boosted to reach line levels. With high input signals, distortion and oscillation, the output will be very high. As as result, settings of an output level potentiometer will vary considerably, according to the effect required. Without IC3B, the range of filter behaviours is sqeezed into a somewhat narrower range, running counter to the philosophy of the design.

Exponential Converter

A simple exponential converter circuit is provided, using readily available components. The basic circuit, with minor additions, comes from a couple of sources: “Electronic Synthesizer Construction” by R. A. Penfold ISBN 0-85934-159-3 page 16, and notes on the Transcendent 2000 music synthesizer, designed by Tim Orr, in the article “Music Synthesizer” published in Electronics Today International, July 1978, page 41.

The current flowing through Q2 is exponentially dependent upon the voltage applied across its base-emitter junction. This voltage results from the mixing of various potentials, including those provided by PR1, a cutoff potentiometer connected to PL3 and optionally a control voltage input at pin 3 of PL4. Q1 provides temperature compensation for Q2, and the faces of Q1 and Q2 can be stuck together to maintain thermal continuity between them. Q3 and Q4 provide a compensated current source for the current-controlled OTAs used as current-controlled resistors in two filter sections.

When the filter module is used on its own, the current for the OTAs is supplied from FC OUT (pin 3 of PL1) to FC IN (pin 2). If the filter is used in a stereo pair, the second module's exponential converter can be left unused, and FC OUT on the first module is connected to FC IN on both modules. Under these circumstances, PR2 on each module can be used to vary the proportion of current supplied to each filter, thereby adjusting the frequency ratio, or the pitch offset. As an alternative, the wiper of a potentiometer can be connected to FC OUT, and its two terminals connected to FC BAL (pin 1 of PL1). The pot can then be used as an external cutoff balance control. The sensitivity of this balance control depends on the potentiometer's resistance; suggested values to try are 5K or 10K. To trim the point at which the two frequencies are the same, connect FC BAL to FC IN, on both modules, and use PR2 to make adjustments. This makes it possible to set the centre position of the frequency balance potentiometer as being the point at which both filters have the same cutoff. This use of a common exponential converter for linked filters avoids temperature drift that would otherwise occur between them. The cutoff may be slightly temperature sensitive, but both filters are affected equally. Perfect filter matching across the entire frequency range is difficult to achieve, however, due to the manufacturing variations in the ratio of C6:C7, and typically a mismatch of a few hertz will be heard at low cutoff frequencies. However, this mismatch is not temperature-dependent.

Ideally, there would be no temperature-dependent drift of cutoff. In filters, as opposed to oscillators, this is not usually a significant issue, so the present exponential converter design avoids difficult-to-obtain parts as much as possible, and doesn't provide the highest level of cutoff stability that is possible. However, with some extra effort, improvements can be made to the design of this exponential converter circuit, and connections can be made to PL1 that exclude the on-board converter, without the need to cut tracks on the PCB.

There are several sources of minor temperature dependence in the existing circuit. The first arises from the fact that Q1 and Q2 are not on the same die, so a significant temperature difference can, in theory, arise between them. In most cases, temperature changes are slow, and this difference will be slight, but nevertheless, this is a source of temperature dependence. It could be reduced using a transistor array with two transistors on the same chip.

A second source of temperature dependence can be compensated if R9 has a temperature coefficient of 3400ppm/K. Such resistors were once manufactured for the purpose of stabilising exponential converter circuits. They can be made from a combination of an 870 ohm copper-wound resistor in series with a 130 ohm metal oxide resistor (note that ordinary wire-wound resistors are made from constantan, not copper). Having constructed such a resistor, the problem then is to ensure thermal connectivity between this resistor and Q1 and Q2.

A third source of temperature dependence arises from the fact that the proportion of base current in Q3 and Q4 varies with Hfe (transistor current gain), which is a temperature dependent characteristic. If the Hfe is high, this variation is reasonably tolerable. This is the reason a darlington pair has been used for Q3 and Q4.

Suggested Experiments

The distortion characteristics of this filter are largely defined by the diodes in the clipping circuit. It is likely that the present components are best for many purposes, but there are no rules. Any diodes can be tried. Start off using the standard 1N4148, as a benchmark, and then try other signal diodes, such as germanium, Schottky, silicon or germanium signal transistors with their bases and collectors joined, power diodes (rectifiers etc.). Using different diodes for D2 and D3 will result in asymmetric distortion, and that may also be interesting. LEDs have a very high voltage drop, making them less likely to be useful.

Another possible change is in the values used for C6 and C7. These have been optimally selected for the current design (they are the same as the original Korg circuit), so start with 3n3 and 1n0. But if you want to experiment, try 2n2 and 1n0, or both the same (which is how the capacitors are configured in the later version of the Korg MS10/20 filter), or different capacitor types, such as ceramic instead of polystyrene (this would result in very subtle differences).

CONSTRUCTORS NOTES

Screening of Cables

When wiring up several modules, it's very tempting to avoid screening all connections, simply because of the amount of time required to make screened connections as opposed to using ordinary wire. Experienced constructors will know when and where it is appropriate to take these short-cuts. However, one particular set of connections is worth examining in detail. The high pass input, PL5.5, is of sufficiently high impedance to inductively pick up a signal from any neighbouring unscreened connections.

A typical scenario is when connections are made to the high pass and low pass inputs using unscreened wire. In this case, the high pass input will receive some signal by induction even when not directly connected. As a result, with the filter supposedly acting on only the low-pass input, some high pass signal will get through. This will be particularly noticable when the cutoff frequency is very low, under which conditions the characteristic muffled sound will be accompanied by some very high frequency content. The solution is to screen the high pass cable.

Conversely, the low pass input, PL5.3, has a lower input impedance, but in any case, low frequencies are less able to inductively couple from neighbouring wires, and so the converse situation, of bleed-through into the low pass input, is unlikely to be audible.

CALIBRATION

Calibration Procedure

Starting with presets PR1 to PR6 at unknown or random settings.

If there are multiple modules in a cascade, set the cutoff and resonance potentiometers (connected to PL1 and PL6 respectively) to minimum. Set PR6 on all modules to minimum. Calibrate the last module in the chain first, then open the filter up with zero resonance, so that the filter preceding it can be heard and calibrated.

If modules are operating in parallel with a shared exponential converter, the cutoff calibration steps should be performed on the module whose exponential converter is being used.

Cutoff Calibration

Set the input level potentiometer to minimum.
Set the output level to about one quarter.
Set the cutoff potentiometer to minimum.
Set the resonance potentiometer to maximum, then reduce so the circuit is just self-oscillating.

Monitor the output signal. Note that signals can be extremely high. The output will easily drive a pair of headphones.

Set PR1 and PR3 to maximum.
Set PR2 to minimum (especially important when using a low voltage power supply, such as ± 9V).

Now sweep PR1 slowly to minimum. The filter should be self oscillating, and the frequency should go from high (or inaudible) to very low.

If no tone is heard, check all connections. Also make sure that there is a connection between pins 2 and 3 on PL1 (i.e., the plug need have no external connection, but can be used solely to connect these two pins).

Once the sweeping tone is heard, continue, ensuring the cutoff potentiometer is set at minimum:

Adjust PR1 from minimum upwards, until a low frequency is heard - this defines the lowest cutoff frequency of the filter.

Now set the cutoff potentiometer to maximum. The oscillation should be inaudible. Reduce PR3 until oscillation is audible again, then increase PR3 slightly, moving it just out of the audible range. This defines the highest cutoff frequency of the filter.

Resonance Calibration

Having calibrated cutoff frequency, continue:

Set the cutoff potentiometer half way. Adjust the resonance potentiometer from maximum to the onset of oscillation (should be just above half way). Now adjust PR5 until resonance begins at the desired position of the resonance potentiometer. PR5 has a limited adjustment range, and its primarily purpose is to match the onset of self oscillation in stereo modules.

Control Voltage Calibration

PR4 is used to adjust the sensitivity of the control voltage input on pin 3 of PL4. Set the filter to self oscillation. Supply a control voltage and adjust PR4 until the desired range of cutoff variation is achieved.

Parallel Cutoff Calibration

This is only needed when parallel modules use the same cutoff control and share a common exponential converter circuit.

Perform the basic cutoff calibration as described above for the module whose exponential converter is being used. Then perform basic resonance calibration for each channel, aiming to obtain the onset of self-oscillation at the same resonance potentiometer setting.

Then, monitoring both channels simultaneously, set the cutoff potentiometer half way, adjust the resonance so both filters are just at self oscillation, and then adjust PR2 on either or both modules to obtain a matching frequency of oscillation.

Note that, due to natural variation in component characteristics, frequency matching will not be precise over the full cutoff range. However, because the control current is derived from the same exponential converter, these inconsistencies are not affected by temperature.

Setting Preamplifier Gain

Using the filter as a low pass filter, set the cutoff to maximum and the resonance to zero. Set PR6 to minimum.

For line level inputs, PR6 can be set to around 15% (“9 o'clock”). Lower level inputs require higher settings of PR6.

As a rough guide, the signal should remain unclipped when the input level potentiometer is set at half way. Above this, distortion should become apparent. When the filter cutoff is reduced, and resonance increased, distortion also increases.

When set up in this way, fairly undistorted sounds are possible with the input level potentiometer set low. The output level needs to be boosted to compensate for such low input levels. However, increasing the input level to maximum (and reducing the output level), results in a very distorted signal, giving the filter a broad continuum of distortion characteristics.

MODULE CONNECTIONS

PL1 – Frequency Control

PL1.1 FC BAL

Frequency control balance. When a pair of modules shares the same exponential converter circuit, this pin can be used to provide a cutoff balance control. PL1.3 FC OUT is connected to the wiper of a 5K or 10K linear potentiometer. PL1.2 and PL1.3 of the same module are connected to one terminal, and PL1.2 and PL1.3 of the parallel module are connected to the other terminal. PR2 on either or both modules can be adjusted so that the central setting of the cutoff balance potentiometer results in matching cutoff frequencies (which are most easily calibrated by causing setting both filters to self oscillate with no input signal).

Different potentiometer values result in different frequency balance ratios (or pitch offsets) being achievable.

PL1.2 FC IN

Receive the frequency control output current from PL1.1 either on the same module, or a module whose exponential converter circuit is being used. Custom exponential converters (for example, an ultra-stable fully temperature-compensated circuit) can supply frequency control current to this pin.

PL1.3 FC OUT

The output of the exponential converter circuit, which can be connected only to PL1.2 on the same board, or in addition, to other boards whose frequency is desired to be the same, or in a fixed ratio.

PL2 – Power Supply

PL2.1 V-

-15V at 25mA (max) 20mA (typical) of a ±15V regulated power supply, or -12V at 20mA (max) 17mA (typical) of ±12V. -9V at 15mA (max) 10mA (typical) for battery operation.

PL2.2 GND

0V power supply connection.

PL2.3 V+

+15V at 25mA (max) 20mA (typical) of a ±15V regulated power supply, or +12V at 20mA (max) 17mA (typical) of ±12V. +9V at 15mA (max) 10mA (typical) for battery operation.

PL3 - Cutoff

PL3.1

The low terminal for the cutoff potentiometer. The recommended value for this potentiometer is 10K ohms linear.

PL3.2

The wiper connection for the cutoff potentiometer.

PL3.3

The High terminal for the cutoff potentiometer.

PL3 is left unconnected in modules that receive their FC IN from the exponential converter circuit in another module.

PL4

PL4.1 INPUT

Input to the preamplifier. 220K ohms impedance. The gain of the preamplifier is adjustable using PR6. An input level potentiometer should be provided on the **output** of the preamplifier, not on this input, so as to reduce unwanted noise.

PL4.2 GND

Ground (for screen connection of INPUT).

PL4.3 CV IN

An external cutoff control voltage can be applied to this pin. Sensitivity is adjusted using PR4. If CV is not required, just leave this pin unconnected.

PL4.4

Ground (for screen connection of CV IN).

PL5

PL5.1 PREAMP OUT

Output from preamplifier, suitable for supplying to an input level potentiometer, suggested value: logarithmic 5K (4K7). When modules are configured in stereo, twin-ganged pots may be used. However, logarithmic 5K stereo pots are hard to obtain, so logarithmic 10K potentiometers may be used instead.

PL5.2 GND

Ground (for screen connection of PREAMP OUT).

PL5.3 LPF IN

Input to filter for attenuating signals above the cutoff frequency. Typically this is taken from the wiper of an input level potentiometer supplied by PREAMP OUT, or from PL7.3 CASCADE. Usually a switch will determine which of LPF IN or HPF IN will receive the input signal (the combination of the two is not very interesting, due to the fact that both HPF and LPF paths are in-phase; making one of these out of phase may resulting in more interesting combined phenomena).

PL5.4 GND

Ground (for optional screen connection of LPF IN).

PL5.5 HPF IN

Input to filter for attenuating signals below the cutoff frequency. Typically this is switched, so that either this or the LPF IN pins receive an input signal. However, even when the switch is only connecting signals to the LPF IN pin, it is possible for a length of wire connected to HPF IN to receive some signal from any neighbouring unscreened wires, resulting in audible high frequency signals even when the filter is being used for low pass only. To avoid this, screen wires connected to HPF IN. An alternative is to only screen the PREAMP OUT connection to the input level potentiometer (as shown in the module configuration schematics), as long as no other high amplitude signal cables run alongside HPF IN.

PL5.6 GND

Ground (for optional screen connection of HPF IN).

PL6 - Resonance

PL6.1

The low terminal for the resonance potentiometer. The recommended value is 10K ohms linear.

PL6.2

The wiper connection for the resonance potentiometer.

PL6.3

The high terminal for the resonance potentiometer.

PL7 – Output

PL7.1 OUTPUT

Main filter output. This has a high signal level (up to 3V RMS), suitable for supplying to an output level potentiometer, suggested value: logarithmic 5K (4K7). When modules are configured in stereo, twin-ganged pots may be used. However, logarithmic 5K stereo pots are hard to obtain, and so logarithmic 10K pots may be used. This output can drive headphones (via the output potentiometer).

PL7.2 GND

Ground (for screen connection of OUTPUT).

PL7.3 CASCADE

This is the same signal as the OUTPUT at pin 1, but with a lower amplitude. Using this terminal, the filter has an overall gain of 1. CASCADE is intended to be supplied to LPF IN or HPF IN of a cascaded module, usually via a switch.

PL7.4 GND

Ground (for screen connection of CASCADE).

PARTS LIST

Preset potentiometers

PR1	20K	PR4	100K
PR2	5K	PR5	5K
PR3	10K	PR6	1M

Resistors

The resistors should have a power rating of 0.25W or more. Accuracy is not critical, so 5% types are probably fine, although 1% is preferred. The construction is preferably metal film, as these seem to result in circuits sounding less harsh than those using carbon film (keep in mind that there are 34 resistors, all contributing a small amount of non-uniform PDF noise).

R1	100K	R13	33K	R25	1K
R2	330K	R14	220	R26	10K
R3	6K8	R15	220	R27	4K3
R4	1K	R16	10K	R28	1K
R5	10K	R17	220	R29	2K7
R6	10K	R18	220	R30	1K
R7	10K	R19	10K	R31	220K
R8	68K	R20	4K7	R32	6K8
R9	1K	R21	750	R33	1K
R10	1K8	R22	1K	R34	10K
R11	6K8	R23	1K		
R12	4K7	R24	27K		

Capacitors

The following three capacitors are not directly in the audio path, and so their construction should not directly affect audio characteristics. Standard ceramic capacitors can be used.

C1	100nF	Ceramic
C2	100pF	Ceramic
C3	100nF	Ceramic

The remaining capacitors are all in the audio path. Electrolytic capacitors are used in several places for audio coupling, but it is felt that any degradation in the performance in these components is a fairly positive part of the sound that is found in old equipment.

The electrolytic capacitors should be at least 35V tolerant.

C4	33uF	Electrolytic
C5	10uF	Electrolytic

C6 and C7 are special: The filter comprises two buffered cascaded RC sections (the 'R' in each section being formed by a transconductance amplifier) and the capacitors in these sections are these two components. Polystyrene capacitors are very much preferred.

C6	3n3	Polystyrene
C7	1nF	Polystyrene

The remaining capacitors are also electrolytic and should be at least 35V tolerant.

C8	47uF	Electrolytic
C9	4u7	Electrolytic
C10	47uF	Electrolytic
C11	22uF	Electrolytic

Diodes

Standard silicon signal diodes are used for D1 to D3. D2 and D3 affect the distortion characteristics. The circuit was designed to use 1N4148's, but experimentation with other types, including germanium, rectifiers, etc. may give some interesting results. However, the circuit is best constructed using 1N4148s to start with.

D1	1N4148	Standard silicon signal diode
D2	1N4148	Standard silicon signal diode

D3	1N4148	Standard silicon signal diode

Transistors

Q1 and Q2 are not in the audio path, and should be high gain (C grade) silicon NPN transistors. A plastic TO92 package is recommended, so that the faces of Q1 and Q2 can be stuck together to conduct heat and keep them at the same temperature. A BC549C meets these criteria.

Q1	BC549C
Q2	BC549C

Q3 and Q4 are silicon PNP transistors.

Q3	BC557
Q4	BC557

Q5 is an audio buffer in the feedback path. The signal levels are usually fairly high, but nevertheless it is probably worth using a low noise high gain transistor such as a BC549C.

Q5	BC549C

Integrated Circuits

IC1 is not in the audio path, but is chosen to be the same as IC3 and IC4 to keep the parts list simple. IC2 is a dual transconductance amplifier.

IC1	LM1458N	Dual 741 op amp
IC2	LM13700N	Dual transconductance amplifier

IC3 and IC4 provide audio amplification in various parts of the circuit. Although these amplifier stages are not intended as primary sources of audio distortion (D2 and D3 take care of that), LM1458s have been chosen because of their distortion characteristics (the LM1458 is basically a dual 741).

IC3	LM1458N	Dual 741 op amp
IC4	LM1458N	Dual 741 op amp

Connectors

PL1	3-way
PL2	3-way
PL3	3-way
PL4	4-way
PL5	6-way
PL6	3-way
PL7	4-way

Additional Components

Several additional components are required in addition to those in the module itself. See the example module configuration schematics for details.

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