The Multiple Identity Filter™, Part I

HERE IS A FILTER STRUCTURE that uses only three ICs, has no commercially available equivalent, is only slightly more costly than standard filter modules that do a lot less, and is amazingly versatile. How versatile? Here are some of the responses obtainable with this module:

Lowpass: Choice of 24 18 12 or 6dB/octave slope; includes voltage-controlled (VC) resonance option in 24dB/octave mode.

Highpass: Choice of 24, 18, 12, or 6dB/octave slope; includes VC resonance option in 24dB/octave mode.

Phase shifter (allpass): Single or dual notch response. VC resonance option in dual notch mode.

Maximum rolloff lowpass: Similar to 24dB/octave lowpass, but with faster rolloff. A typical application would be getting rid of clock feedthrough from analog delay line outputs.

Notch filter: Choice of broad or sharp notch response.

Bandpass: Choice of sharp or broad bandpass response.

Lowpass plus notch: Fig. 1 shows the response curve for this type of filter.

Highpass plus notch: Another new type of response. See Fig. 2.

Since this module can assume the identity of many other filter structures, the term "Multiple Identity Filter" (MIF for short) seems very appropriate. But there's still more; it can act as a stand-alone signal phaser, and can put different filter blocks in parallel for highly unusual responses (for example, you could have a I2dB/octave lowpass filter in parallel with a single notch phase shifter, or you could parallel four phase shift sections). There's even a straight channel option with switchable polarity (more on this later). And it's also a sine wave oscillator....

Sound incredible? Well, I still haven't discovered all the possible sounds obtainable from this module, so I don't think you'll get tired of it too easily. You do pay a price for this versatility, the time required to learn how to use the device to best advantage. However, you can always treat the thing like a standard lowpass filter, and learn about the many extras later on at your own pace.

About the circuit. The MIF is based on the CEM3320, a new 4-pole filter IC from Curtis Electromusic Specialties [2900 Mauricia Ave., Santa Clara, CA 950511]; but note that the MIF structure is also equally applicable (with some minor design changes) to the SSM2040 filter IC. The CEM3320 contains four 6dB/octave filter blocks, along with a resonance VCA to give voltage-controlled resonance. There is nothing new about configuring different filter sections to give different responses (see the CEM3320 data sheet, the SSM2040 data sheet and applications notes, and the article "Blacet 'Phasefilter' Review" in Device newsletter, Vol. 1, No. 6 [Box C, Carmichael, CA 95608]), but the way in which it is done here is unique. Coupling these programmable filters with switchable, series/parallel patching that's integral to the module is the key to giving the MIF its unprecedented versatility.

Filter Block Characteristics. Fig. 3 shows the MIF's block diagram. Each filter section has two switches that program the response of the filter (highpass, lowpass, allpass, or disabled). Each filter block also has a switch to choose between two different timing caps, placed a decade apart. This allows for unusual slopes, changing the resonant frequencies of filters connected in parallel, and so on.

Series/Parallel Switching. The series/parallel switching scheme has been previously described elsewhere ("Generalized Series/Parallel Switching" in Device, Vol. 1, No. 7). With all three switches in the S position, the four filter blocks are in series. With the switches in the P position, all four blocks are in parallel. Fig. 4 shows some possible configurations, along with the switch settings required to produce these results. Note that when the input of any filter connects to the output of the input buffer (parallel mode), then the output of the preceding filter stage connects to the output mixer. I realize that the switchable patch scheme may be confusing at first—but trace the circuit through for various settings of S1, S2, and S3; it should all make sense after about 15 minutes of checking it out.

Resonance Cell. A resonance VCA connects from the output of the last filter section to the input of the first filter section, and is therefore most usable when all four filters are in series. It has a panel control that sets the resonance, and a control voltage input jack. Plugging into the jack disables the from panel control and allows for a 0 to +10V control of resonance (with high resonance, the filter changes identity to a sine wave oscillator).

Input Buffer. This section conditions signals going to the filter, and has three inputs. One input is noninverting; one is noninverting and includes an attenuator; the remaining input is inverting. Phase compatibility with parallel filters is very important, and having a choice of inputs allows you to choose your phase.

Straight Channel. A fader allows you to mix unfiltered sound in with the filtered sound. A phase switcher in the straight signal path allows you to choose the straight signal phase.

Control Voltage Amp. This has two inputs that follow the industry standard, 0 to +10V, exponential response control curve. One input includes an attenuator, the other does not. There is also an initial frequency control.

Part I Conclusion. I realize the all this may be confusing—after all, programmable filter structures, integral switchable patching, and optional phaser response are not part of the "standard" synthesizer lexicon (but if I get my way, they will be soon!). I do think that after the conclusion of this series, though, everything will fall right into place. In any event, I hope I've managed to convey some of my excitement for this device, and I hope you're excited about it too. Next month, it's construction time.

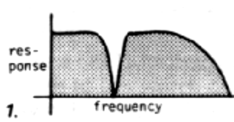


Fig. 1.

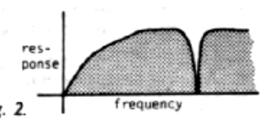


Fig. 2.

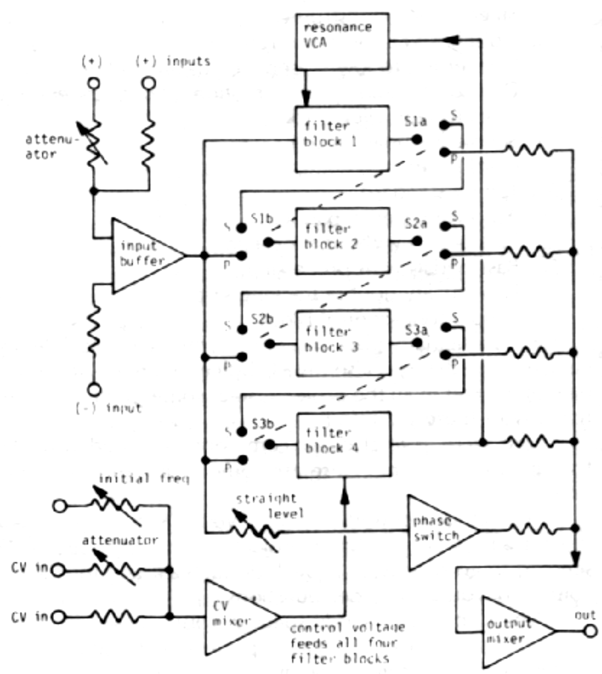


Fig. 3.

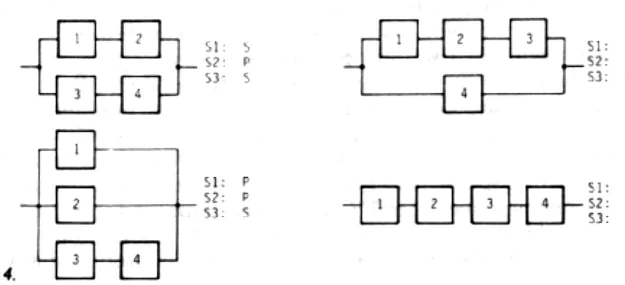
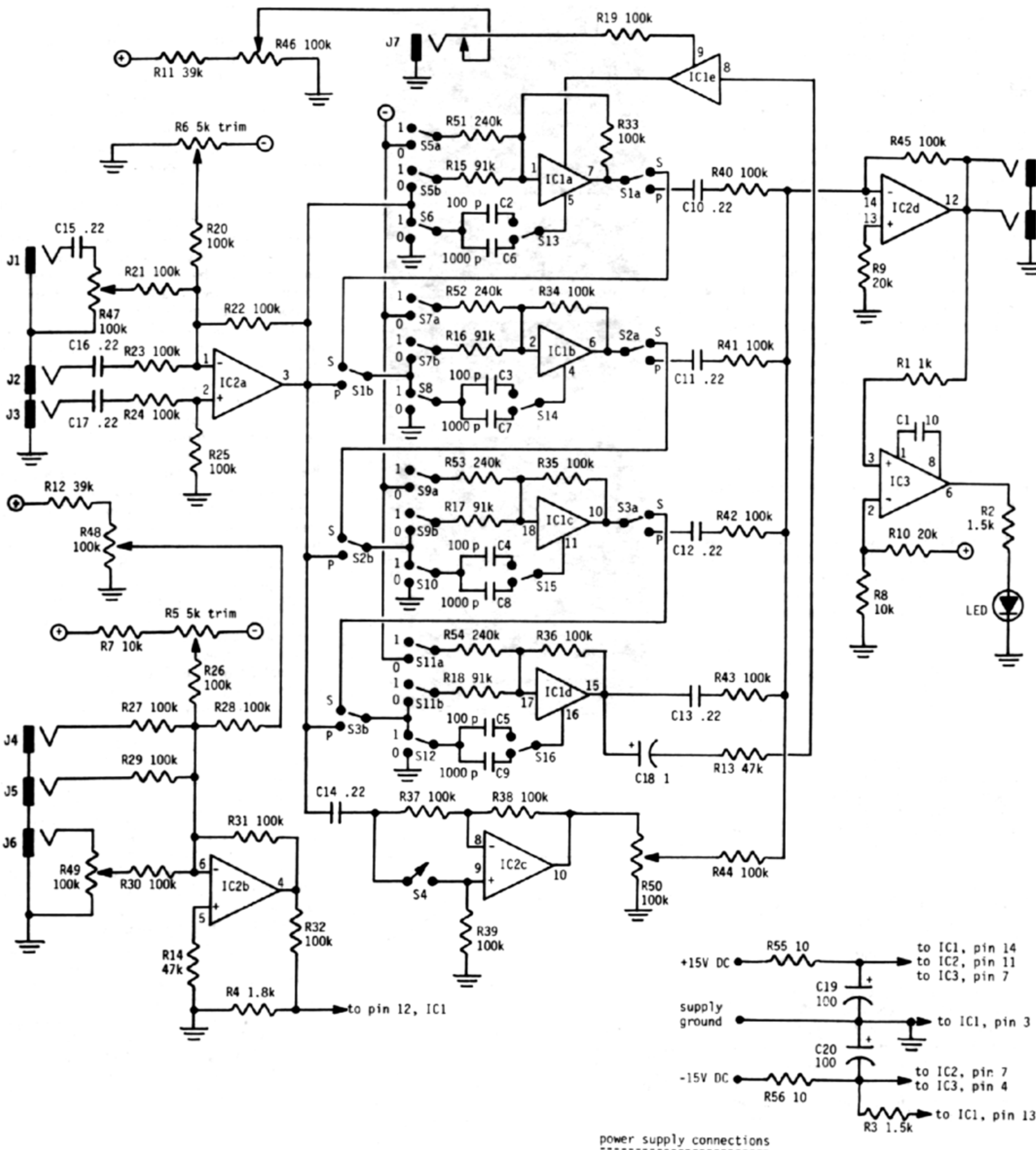


Fig. 4.

Multiple Identity Filter™, Part II

THIS MONTH, THERE'S ONLY ROOM for the schematic and the annotated parts list. Next month, we'll give construction tips and calibration procedures. Please do not attempt to wire this beast up until then, *no matter how experienced you are;* the panel layout and switch orientations are just as important as the circuitry, and you'll want to wait until that information is presented before you actually start building. We'll have some photos of a prototype built from the parts kit, which should also help in building this module.



CRAIG ANDERTON PARTS LIST

Resistors (all are 5%, 1/4 watt units except for the pots)

R1	1k
R2,R3	1.5k
R4	1.8k resistor. For maximum temperature stability use a +3300 ppm temperature compensated resistor (Tel Labs Q81). To use the 2k tempco resistor offered by the Electronotes parts service, change R32 to 110k. You can always wire in a regular resistor first, and change it over later to the temperature compensated kind.
R5	5k trimpot. Adjusts control voltage (CV) amp offset. See calibration instructions next month.
R6	5k trimpot. Adjusts output of 1C2a to +6.5V DC with no input signal. See calibration instructions next month.
R7,R8	10k
R9,R10	20k
R11,R12	39k
R13,R14	47k
R15-Rl8	91k
R19-R45	100k
R46	100k linear pot. Controls resonance when J7 is not used for an external control voltage.
R47	100k audio or linear taper pot. Use to attenuate audio input signal If clipping indicator (LED) starts to glow.
R48	100k linear pot. Sets initial filter frequency (offset).
R49	100k linear pot. Attenuates CV source plugged into J6.
R50	100k audio or linear taper pot. Controls level of straight audio signal.
R51-R54	240k
R55,R56	10 ohms

Capacitors (all should be rated to work at 15V or more)

C1	10 to 20pF ceramic disc
C2-C5	100pF; use polystyrene cap for best stability.
C6-C9	1000pF; use polystyrene cap for best stability.
C10-C17	.22μF mylar or disc
C18	1μF tantalum or electrolytic
C19,C20	100μF electrolytic

Semiconductors

IC1	Curtis CEM3320 [Curtis Electromusic Specialties, 2900 Mauricia Ave., Santa Clara, CA 95051. Hobby market distributor: PAIA Electronics, 102D W. Wilshire Blvd, Oklahoma City, OK 73116. Cost is $8.95 in single quantities; add $1 postage/handling].
IC2	RC4136, XR4136, or equivalent quad low-noise op amp.
IC3	LM748 or LM301 uncompensated op amp.
LED	Any red LED will do just fine.

Mechanical parts

J1	1/4" mono open-circuit phone jack. Accepts an audio input signal that may be attenuated by R47.
J2,J3	1/4" mono open-circuit phone jacks. These jacks accept additional audio input signals if desired.
J4,J5	1/4" mono open-circuit phone jacks. These jacks accept any 0 to +10V control voltage to control the filter frequency.
J6	1/4" mono open-circuit phone jack. Accepts a 0 to +10 CV that may also be attenuated by R49.
J7	1/4" mono closed-circuit phone jack. Accepts a 0 to +10 CV to control the filter resonance, and simultaneously disables R46.
J8,J9	1/4" mono open-circuit phone jacks. These jacks provide dual audio outputs from the filter.
S1-S3	DPDT switches that provide series/parallel switching for the various filter sections.
S4	SPST switch. Changes phase of straight channel; when shorted, phase is noninverting. When open, phase is inverting.
S5,S7,S9,S11	DPDT switches, used in conjunction with S6, S8, S10, and S12 to program the responses of the various filter blocks.
S6,S8,S10,S12	SPDT switches, used in conjunction with the above switches to program the responses of the various filter blocks.
S13-S16	SPDT switches. These select the timing capacitor for each filter section.
IC sockets	You will need one 18pin socket, one 14pin socket, and one 8pin socket.

Multiple Identity Filter™, Part III

IF YOU'VE READ **Parts I and II** of this series, you'll be itching to get into the construction of the Multiple Identity Filter (MIF), which I gave the schematic of last month. So let's get to it.

Construction. The photo in fig. 1 shows a prototype module I built on a 9" x 3" Plexiglass panel. The control layout is as follows: The upper row; left to right, comprises the clipping indicator LED, S5, S6, and S13. The row of switches below is S1, S7, S8, and S14, again left to right—in fact, we'll just assume that all locations are given from left to right to minimize repetition. The next lower row of switches is S2, S9, S10, and S15; and the bottom row of switches is S3, S11, S12, and S16. The top row of controls is R50, S4, and R48, while the bottom control row is R47, R49, and R46. The top row of jacks is J1, J6, and J7; the row below, J2, J4, and 18; the bottom row, J3, J5, and J9.

Switches S1-S3 should be oriented during construction so that toggling them upwards puts them in the "S" position (see last month's schematic), and switches S5-S12 should be oriented so that toggling them upwards puts them in the "1" position. S13-S16 should be oriented so that the upwards position selects the 100pF timing capacitor, while the downwards position selects the 1000pF timing capacitor.

By the way, a parts kit for this module will be available soon from PAIA Electronics, although pricing and delivery are not firmed up at this time (columns are prepared months in advance of publication dates).

Calibration. This process isn't difficult as long as you have a voltmeter. Connect one probe to pin 3 of IC2, and the other probe to pin 10 of IC1; then set the range selector switch for about a 0 to 5V DC. After applying power and setting the initial frequency control halfway, adjust R6 for a 0-volt reading on the meter. You might find adjustment is a bit less critical if you change R20 to a 220k resistor instead of the 100k value given last month, although in any case the module should be calibratable.

There are a number of ways to calibrate R5, all of which are difficult to write down on paper but obvious when you have the unit sitting in front of you. The basic idea is to adjust this control so that a 0V control voltage (CV) gives the lowest desired filter frequency, and a +10 CV gives the highest desired frequency. One calibration procedure for R5 is to feed a low-frequency (approx. 80Hz) square wave into J1, and patch up the MIF for a basic 24dB/ octave lowpass configuration (see patch chart). Vary the initial frequency from lowest to highest frequency (0 to +10V), and leave it at the highest frequency position. Set R5 to the extreme of rotation that gives the *most* high frequencies. Now slowly rotate R5 towards the other extreme until you hear just the slightest reduction in high frequency content. Leave R5 right there,

PHOTO BY VESTA COPESTAKES

Fig. 1.

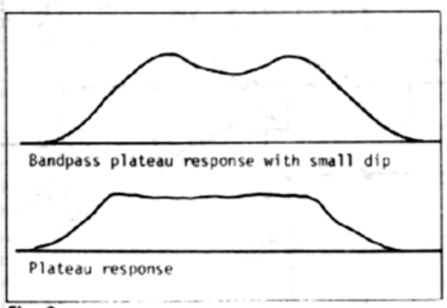


Fig. 2.

and the MIF is calibrated.

Applications. Ask an engineering-minded synthesist what a filter does. and that person will probably tell you about how it alters the waveform of a VCO fed into its input, and how it can change the harmonic structure of a complex waveform. However, ask a recording engineer what a filter does, and the answer **will be** that filters are put together in systems to make up equalizers that alter frequency response, correct for response anomalies, and so on. This difference in operating philosophy has meant that most filters designed for synthesizers perform poorly with program material or when processing acoustic instruments, whereas studio equalizers work well for processing but are generally not voltage-controlled and are used more for subtle corrections than for special effects. The strength of the MIF is that it feels at home in both roles; hopefully, synthesists will get into more subtle types of sound with this module, and conversely, I hope that recording enthusiasts find out that having dynamic—rather than static—equalization can do some *really* amazing things.

If you know how to use a scope and a signal generator, looking at the MIF output for various controls will show you a tremendous amount about how this thing works,. Of course, your ears will tell you the most, but if you can analyze what the MIF is doing electrically, you'll know how to apply it to solve specific response problems that crop up.

There are only a few rules to consider when using the filter. First, when sections are switched in parallel, you'll have to back off on the input signal to avoid clipping (the clipping indicator I should show this condition). Next, the voltage-controlled resonance feature may or may not do something interesting; twist the initial resonance knob and if you get a good sound, great. If nothing happens, that simply means you're using a patch where the resonance cell is inoperative. Another point to remember is that there are many, many responses of which this thing is capable, and it's also possible to duplicate the same response with different patches (for example, there are four or five ways to create a notch response). Finally, there are some patches where you may have to boost the input signal a bit if you're doing a particularly strange patch.

Final Comments. The mini-manual on the opposite page should get you *started* on the road towards using the MIF in a predictable and efficient way; remember, it's going to take you some time to master it. If you discover any particularly good patches or applications with this beast (e.g., "The highpass plus notch response works great with electric pianos"), write me c/o CK and I'll pass these ideas along. And the future? Why, using individual voltage-controlled filters for each MIF section in conjunction with microprocessor control and a little memory.... But in the meantime. I'll try to master the module I have now.

Multiple Identity Filter Mini-Manual

The following diagrams show the orientation of the switches in the switching matrix (S1-S3, S5-S16) for various filter modes; U means up, D means down. Don't forget that in most cases, fooling around with the capacitor select switches (S13-S16) will create new responses. Adding in some straight signal, either in phase or out of phase, will create other changes. The designation X means the switch position is irrelevant.

Lowpass, 24dB/octave
D D U
U D D U
U D D U
U D D U

Highpass, 24dB/octave
U U D
U U U D
U U U D
U U U D

Lowpass, 18dB/octave
D D U
U D D U
U D D U
D U D X

Highpass, 18dB/octave
U U D
U U U D
U U U D
D U D X

Lowpass, 12dB/octave
D D U
U D D U
D U D X
D U D X

Highpass, 12dB/octave
U U D
U U U D
D U D X
D U D X

Lowpass, 6dB/octave
D D U
D U D X
D U D X
D U D X

Highpass, 6dB/octave
U U D
D U D X
D U D X
D U D X

Bandpass, 12dB/octave slope
D D U
U D D U
U U U D
U U U D

*Add (-) straight signal for inverted (notch) response.
*Add resonance to produce plateau response (see Fig. 2)

Bandpass, 24dB/octave slope
U U D
U D D U
D U D X
U U D X

*Add (+) straight signal for inverted (notch) response.

Lopsided bandpass (low frequency slope 6dB/ octave, upper frequency slope 18dB/octave)
D D U
U D D U
U D D U
U U U D

*Add (+) straight signal for inverted response (lopsided notch).

Lopsided bandpass (low frequency slope 18dB/ octave, upper frequency slope 6dB/octave)
D D U
U U U D
U U U D
U U U D

*Add (+) straight signal for inverted response (lopsided notch).

Bandpass Plateau response with small dip (see Fig. 2)
U U D
U D D D
D U U U
U D D U

Highpass plus notch response (also lowpass plus notch)
D U U
U D U U
U D U U
D U D X

*Add in straight signal for HP plus notch.
*Add in straight signal for LP plus notch.

Allpass, dual notch
D U U
U D U U
U D U U
U D U U

(notches spaced approximately 6:1 apart with respect to frequency)

D U D
U D U U
U D U U
U D U U

(notches spaced approximately 10:1 apart with respect to frequency)

D U U
U D U U
U D U D
U D U D

(notches spaced approximately 16:1 apart with respect to frequency)
*Add in(+) straight signal to create dual notch.
*Add in (-) straight signal to reverse position of peaks and notches.

Allpass, single notch
D U U
U D U U
D U D X
U U D X

*Add in (+) straight signal to create single notch.
*Add in (-) straight signal to create bandpass response.

Single notch
U U D
U U U U
D D D U
U D D U

(shallow notch—about -6dB)

U U U
U U U U
D D D U
U D D U

(deep notch—about 3OdB)