FILTER BANK DESIGN:

-by Ian Fritz

I. Introduction

In this article a fixed frequency filter bank design that has been in use for the past couple of years will be described. The original impetus for this project came from the work of Mathews [1] and of Burhans [2] who described filter bank techniques for timbral enrichment. Those authors suggested using logarithmically distributed resonant frequencies with about five filters per frequency octave (distributed through the middle audio range) and with peak-to-valley ratios of about 10 db. The unit described below is based on the set of resonant frequencies suggested by Burhans [2], but has more flexibility than the original "timbre box" design in that the output amplitude of each filter is individually adjustable and in that the Q's of the filters are adjustable within groups of frequencies. These adjustments make a wide variety of frequency responses possible, so that the filter bank is useful for fixed format modi-fication as well as for timbral enrichment. The actual design is a bit of a monstrosity and takes a lot of work to build and adjust. As will be discussed below there are some slightly quirky features to the operation of the device, which could probably be fixed with a better design. However, given the continuing interest in filter banks [3], it was felt worthwhile to present the design and some comments of the use of the unit.

II. Overview

The thirty-six resonant frequencies suggested by Burhans are listed in Table I, along with the frequencies of the tempered chromatic scale. Tuning of the filters may be done by setting the frequencies on or near the appropriate notes or "cracks". This makes it possible to set up the filter bank without a frequency counter if a keyboard instrument is available. The frequency ratio of adjacent filters is $\sqrt[5]{2.1}$.

A block diagram of the system is given in Figure 1. The filters are all in parallel, but arranged in groups of six (subbanks). Each subbank has variable feedback for Q variation: thus each group of six can have a different Q. At maximum Q settings the response is the same as the Burhans timbre box, i.e., Q's of $\approx$ 20 and 10 db peak-to-valley variation. At minimum Q the ripple is only $\approx$ 3 db, and much smoother in variation. The amplitude controls marked "A" in Figure 1 are to compensate for the change in overall gain caused by variation of the Q controls.

III. Circuit Details

Details of the circuitry will be described from the inside out, i.e., starting with the individual filters. Two prime concerns in designing the filter bank were cost and noise performance. It was anticipated that a unit based on inexpensive op amps might be excessively noisy, so a discrete circuit was used. (Now a reasonably

Fig. 1
Block Diagram

Table 1. Concert Pitch Frequencies (Freq.) and Filter Resonant Frequencies (Filter)
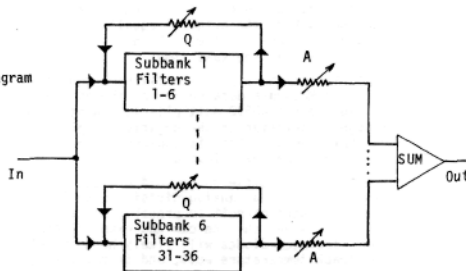
PITCH	Freq.	Filter	Freq.	Filter	Freq.	Filter	Freq.	Filter	Freq.	Filter	Freq.	Filter	Freq.	Filter	Freq.	Filter
G	49.0	50	98.0		196		392	398	784		1568		3136	3186	6272	
G#	51.9		104	105	208		415		831	838	1661		3322		6645	6690
A	55	58	110		220	221	440	463	880		1760	1760	3520	3696	7040	
A#	58.3		117	122	233		466		932	972	1865		3729		7459	7762
B	61.7		124		247	256	494		988		1976	2042	3951		7902	
C	65.4	67	131		262		523	537	1047		2093		4186	4288	8372	
C#	69.3		139	141	277		554		1109	1128	2217		4435		8870	9005
D	73.4		147		294	295	587	621	1175		2349	2369	4699	4974	9397	
D#	77.8	78	156	164	311		622		1245	1303	2489		4978			
E	82.4		165		330	344	659		1319		2637	2736	5274			
F	87.3	91	175		349		699	722	1379		2794		5588	5746		
F#	92.5		185	190	370		740		1480	1517	2960		5920			

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inexpensive low-noise option is available in the 739 dual op amp). The basic circuit, shown in Figure 2, is built around a high gain (β > 300) low noise transistor (Q1, 2N5088). Positive feedback is provided by the phase-shift network formed by C3, C4, C5, R4, R6 and R7. The circuit would be a standard phase-shift oscillator if the emitter were held at ac ground (C2 large, TP1 = 0Ω). As it is, TP1 effectively varies the amount of feedback, producing a variable Q filter design. The values of components not listed in the figure are different for different sections, and are tabulated in Table II. Values of C3, C4 and C5 given are only approximate, and are values calculated from an emperically derived formula. The frequencies of the filters are adjusted by trimming these capacitors.
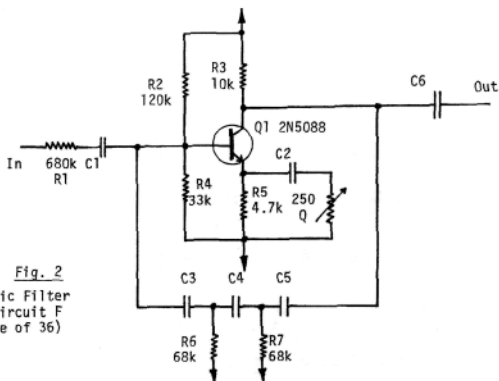
Fig. 2
Basic Filter
Circuit F
(one of 36)
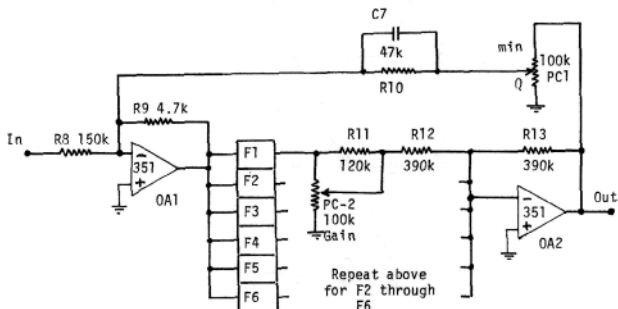
Fig. 3 Interconnection into Subbanks SB of six (one of six)

Table 2 Capacitor Values

FREQ.	C3	C4=C5	C1	C2	C6	C7
50	0.036μ	0.018μ	0.03μ	250μ	0.3μ	0.01μ
58	0.031	0.015				
67	0.027	0.013				
78	0.023	0.011				
91	0.020	0.010				
105	0.017	0.0084				
122	0.015	0.0072	0.013	100	0.13	0.0047
141	0.013	0.0062				
164	0.011	0.0054				
190	0.0095	0.0046				
221	0.0081	0.0040				
256	0.0070	0.0034				
295	0.0061	0.0030	0.0047	50	0.047	1800p
344	0.0052	0.0026				
398	0.0045	0.0022				
463	0.0039	0.0019				
537	0.0034	0.0016				
621	0.0029	0.0014				
722	2500p	1220p	2200p	22	0.022	1000
838	2150	1050				
972	1850	910				
1130	1590	780				
1300	1380	650				
1520	1180	580				
1760	1070	500	1000	10	0.01	680
2040	880	430				
2370	710	370				
2740	560	320				
3190	510	280				
3700	470	240				
4290	420	210	390	4.7	0.005	400
4950	370	180				
5750	320	150				
6690	270	130				
7760	230	115				
9000	200	100				

The interconnection of the sets of six filter circuits into subbanks of six is shown in Figure 3. At this point some of the quirks of the unit become apparent. Since each filter section has a high gain, the input must be highly attenuated. The combination of R8 and R9 serve this function. Each filter has an attenuator network R11-PC2 on its output, with PC2 being the gain control for the filter section in question. The outputs are summed by 0A2. It works out that the Q of the filters depends on the impedance of the output network. Resistor R11 is included so that the Q decreases slightly as the gain is reduced. Without R11, the Q increases and the filter breaks into oscillation at low gain. The feedback network R10, C7, PC1 is used for Q variation of all the six filters in the subbank simultaneously. Note that the feedback is degenerative: the individual filters are adjusted for high Q ($\simeq$ 20) with no feedback, and the Q is reduced as feedback is added. It was found that feedback does not affect all the filters the same - the lower frequencies are attenuated more than the higher ones. The reason for this is not known, but the effect can be reduced considerably by the inclusion of C7. Values of C7 are tabulated in Table 2.
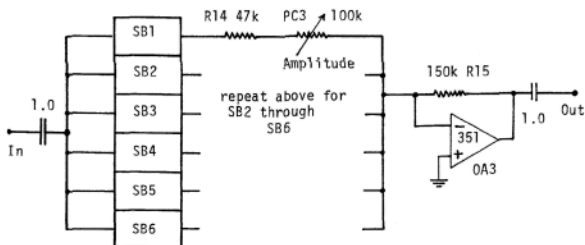
Fig. 4 Interconnection of Subbanks

The interconnection of the subbanks into the complete unit is shown in Figure 4. The "amplitude" controls PC3 are used to compensate for the loss of gain in the subbanks when the Q is reduced; thus PC1 and PC3 are normally varied together.

IV. Adjustment

Adjustment of the filter bank consists in setting the proper Q's with the trimpots TP1 and in adjusting the frequencies by varying the capacitors C3, C4 and C5. The adjustments interact slightly, so it is best to set the Q's first, then fix the frequencies and finally trim the Q's again. A keyboard controlled VCO and an oscilloscope can be used for the adjustment procedure. It is easiest to work with small sections of the bank at a time: this can be done by turning up the gains of, say, five adjacent sections, turning down the rest, and adjusting the middle three of the five. When adjusting a section be sure the adjacent sections are turned up. The bank should be set up for the "timbre-box" mode, as this requires the most critical adjustment. Start with the panel Q controls (PC1) at maximum and with PC3's at minimum. Set all the trimpots at minimum Q and be certain no sections are oscillating. If they are it may be necessary to add a few hundred ohms in series with their trimpots. Next turn up the trimpots one at a time while manually sweeping the VCO through the resonances, and adjust the Q's so that the peak-to-valley ratio's are about 3-4 (10-12 db). This will only be a rough adjustment, since the filter frequencies will not have the proper spacing. Now the unpleasant part - adjusting the frequencies. Use Table I and the keyboard (tuned correctly) to find out if the resonances are in the right place. Turn up only one section at a time to do this. Adjust one or more of the three caps by adding series or parallel trimmers until the frequency is in the right place. Make sure that the construction layout is such that this can be done easily. (Also make sure you have a huge selection of caps!) Once the frequencies are set, readjust the Q's for the 10 db peak-to-valley ratios. Fine trimming of the frequencies to obtain uniform spacings may be necessary. Finally, check the low Q operation of the bank by turning the six panel Q's up and the six amplitude controls down. The response won't be perfectly flat, but the variations should be relatively smooth. Changing the values of the C7's may be necessary.

V. Operation and evaluation

The filter bank is not a precision device in the sense that the peaks and valleys (in the timbre-box mode) are not perfectly uniform and in that the response is not perfectly flat in the low-Q mode with all the gains turned up. However, since the purpose of the unit is to deliberately produce a non-uniform frequency response, this lack of precision and uniformity is more of an aesthetic nuisance than a practical hinderance. In fact, the unit works perfectly well for timbral enrichment and spectral modification. Actually it may even be practical to use the quirks of the system (interaction of gain and Q controls with each other and with resonant

frequencies) to advantage. For example, Tomita [4] has discussed his technique for producing string-section sounds by overdubbing tracks recorded through a filter bank with slightly different filter bank responses for each track. Small changes in the filter-bank response are easily made with the present unit.

Although the operation of the filter bank is satisfactory, there are still some interesting unanswered engineering questions concerning the design and there is still room for design improvement.

One of the main engineering concerns when 36 circuits are built in parallel has to be noise. The 2N5088 is a good low-noise device, but the filter sections have a large gain (to overcome the losses in the feedback networks) so they do produce some noise. Is there a simple transistor circuit that would require less gain? How does the noise of the present circuit compare with that of an op amp device? It would be interesting to hear from any expert readers who could answer these questions. Noise gates can be added at the outputs of the subbanks to turn off the noise when there is no signal in the corresponding frequency band. A working design is given in Figure 5. This circuit is to be inserted immediately following OA2 in Figure 3.
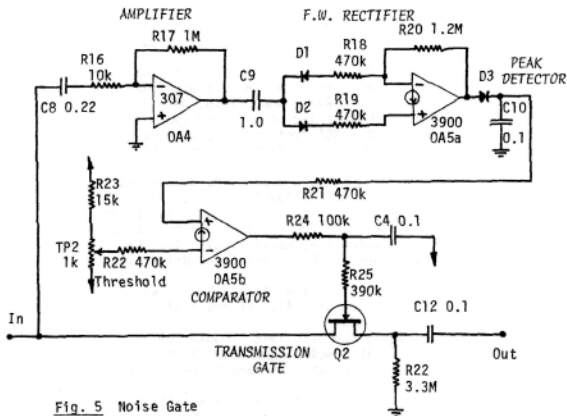
Fig. 5 Noise Gate

The design described in this article is flexible in the kind of responses it produces, but it also has a lot of controls to adjust. A somewhat simpler design might still have enough flexibility for practical applications. For example, the 36 gain controls could be replaced by on-off switches: if you wanted to emphasize a single frequency it would be easier to flip 35 switches than to turn down 35 pots. Also each pair of Q and A controls could be replaced with a single two or three position switch. In general, however, the design has proven versatile and useful as it stands.

References

[1] M. V. Mathews and J. Kohut, "Electronic Simulation of Violin Resonances", J. Acoust-Soc. Am. 53, 1620 (1973).

[2] R. W. Burhans, "Low Cost Comb Filter Methods", J. Aud. Eng. Soc. 22, 335 (1974); "Audio Engineering Improvements for Clavichords", J. Aud. Eng. Soc. 23, 635 (1975).

[3] B. A. Hutchins, "New Ideas on Filter Bank Design", EN#104 (7) (Aug. 1979)

[4] I. Tomita, interview in Synapse 2, 17 (Nov/Dec 1977).

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Questions: Cont. from pg. 2

effects and richer sounding chords.   Unfortunately, this is not really an invitation to be sloppy.   As it turns out, we will probably have to make one of our best efforts at stability, and that which we can't control after that may be just about right. So what you say above is true to a degree.   Separate oscillators are a bit difficult, but a top octave generator is too exact, at least for many applications.

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