Tunable Bandpass Filters in Music Synthesis 4098 (L-2)

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TUNABLE BANDPASS FILTERS IN MUSIC SYNTHESIS

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Abstract:

Subtractive synthesis, or source-filter synthesis, is a well known topic in electronic and computer music. In this paper a description is given of a flexible subtractive synthesis scheme utilizing a set of tunable digital bandpass filters. Specific examples and applications are presented for realtime subtractive synthesis of singing and other musical signals.

0. INTRODUCTION

Subtractive (or source-filter) synthesis is used widely in electronic and computer music applications. Subtractive synthesis generally involves a source signal with a broad spectrum that is passed through a filter. The properties of the filter largely define the shape of the output spectrum by attenuating specific frequency ranges, hence the name subtractive synthesis [1].

The subtractive synthesis model is appropriate for the wide class of physical systems in which an input source drives a passive acoustical or mechanical system. Many musical instruments operate on this principle. For example, a trumpet player's buzzing lips provides the input signal to a passive acoustical filter consisting of the tubes, valves, and bell of the instrument. Similarly, the vowel sounds of human speech or singing are generated by a source of acoustical pulses (lungs and glottis) applied to the throat, mouth cavity, and nasal passages. The shape of the resulting spectrum--and thus the vowel sound produced--is controlled by changes in the dimensions of the vocal apparatus.

In this paper a source-filter synthesis structure employing a set of tunable parametric bandpass filters is presented. The proposed form of each filter follows the Regalia and Mitra design, allowing the center frequency, gain, and bandwidth of each filter to be adjusted individually with single coefficients [2]. This method has a computational advantage over other variable bandpass designs since it is not necessary to recalculate several filter coefficients at once. Moreover, the filters remain stable for all parameter settings.

1. SUBTRACTIVE SYNTHESIS MODEL

A block diagram of a conventional subtractive synthesis system is shown in **Figure 1**. Performing subtractive synthesis requires specification of the source excitation signal and of the desired spectral properties of the filter structure. If a particular physical musical instrument is to be modeled it is sometimes possible to derive the source and filter specifications analytically from the geometry and composition of the instrument itself or from an analysis of the acoustical signal produced by the instrument. In practice the source signal is usually a periodic impulse train (or "buzz") to simulate the spectrally rich characteristics of common musical sources. It is also very desirable in practice to use a low-order filter model so that the computational load is tractable.

The time-varying behaviors of the source and filter are important for a musically interesting result. Singing synthesis, for example, requires a continuous variation in the center frequency of vocal resonances (formants) if vowel transitions and diphthongs are to be synthesized (e.g., the word "loud" has the vowel change "ahh - ooo"). Signal frequency characteristics such as vibrato and glissando are also very important for a successful synthesis technique.

The basic subtractive synthesis structure can also be used for *cross-synthesis* effects. In cross-synthesis the simple source excitation signal is replaced by a signal from some other musical instrument or ensemble. The resulting output of the subtractive synthesis process contains the characteristic timbre of the input source as well as the subtractive spectral shaping of the filter bank [3].

2. BANDPASS EQUALIZER FILTERS

The filter proposed for use in this paper is the second-order bandpass equalizer structure of Regalia and Mitra [2]. The basic filter structure is depicted in **Figure 2**. The portion of the filter labeled A(z) is a low-sensitivity all-pass (delay) function suitable for implementation with a lattice filter structure. Although it is not utilized here, it should be noted that the alternative structure of harris and Brooking [4] with fixed bandwidth properties could be used in a similar manner.

The mathematical representation of the second-order equalizer is given by [2]

$$F(z) = \frac{1}{2}[(1+K)+(1-K)A(z)], \tag{1}$$

where the all-pass section is defined

$$A(z) = \frac{z^{-2}+b(1+a)z^{-1}+a}{1+b(1+a)z^{-1}+az^{-2}}. \tag{2}$$

The Regalia and Mitra equalizer filters have several notable properties that are useful in subtractive synthesis systems. First, the three parameters of the filter (*a*, *b*, and *K*) are functionally related to the bandwidth, center frequency, and boost/cut properties, respectively, of the filter frequency response. Second, the filter parameters are essentially independent, meaning that a change in one parameter does not greatly alter the frequency response features controlled by the other parameters. Third, the frequency response approaches unity gain at frequencies away from the boost/cut range, thereby allowing convenient cascade structures of independent bandpass units. Finally, the filters are stable, minimum phase, and quite insensitive to coefficient quantization.

The basic design relationships are:

Center Frequency ω_0:

$$b = -\cos\omega_o \tag{3}$$

Filter Gain at Center Frequency ω_0:

$$K = F\left(e^{j\omega_o}\right) \tag{4}$$

Nominal Bandwidth Ω (-3dB bandwidth when K=0):

$$a = \frac{1-\tan(\Omega/2)}{1+\tan(\Omega/2)} \tag{5}$$

The influence of the bandpass parameters is shown in **Figure 3** for several values of *a*, *b*, and *K*. Note that the filter response approaches unity gain (0 dB) at low and high frequencies and peaks or dips by the factor K at the center frequency of the filter.

3. SINGING SYNTHESIS MODEL

Several methods for digital synthesis of the singing voice have been demonstrated, including the source-filter model approach used here [5-7], physical models of the vocal tract [8], and the Formant-Wave-Function method [9, 10]. In all cases the choice of parameters is the vital ingredient to the perceptual success of the synthesized signal.

The basic singing synthesis model for vowel sounds used here is shown in **Figure 4.** A periodic pulse generator and lowpass filter are used to simulate the glottal excitation used to drive a set of bandpass equalizer filters adjusted to the desired formant frequencies [1, 6]. Only the boost mode ($K > 1$) of the filters is used. Provision for an additional output filter to simulate the high frequency properties of the vocal apparatus could also included in the model, as could an amplitude envelope control.

Examples of the composite filter spectra are given in Figure 5.

The production of static vowel spectra using the source-filter model is useful analytically, but the application of dynamic spectral variations is of greater creative importance. The use of parametric equalizer filters in the formant model allows gradual changes in each formant frequency, bandwidth, and level in order to create realistic vowel transitions.

The vowel synthesizer employed in this study contains five bandpass filters corresponding to the first five vocal formants. A lookup table of center frequencies, levels, and bandwidths and the corresponding values of a, b, and K for several vowel sounds is used to control the synthesis process. The initial values of the filter parameters are ramped to the final values over a specified duration. The current system uses linear parameter changes for simplicity, although a lookup arrangement using the transcendental functional relationships of Equations (3) - (5) could also be used.

As is the case with most synthesis systems the major obstacle to high quality singing synthesis is the determination of appropriate control parameters and parameter timing for the model. A synthesis-by-rule approach has been used here with some success, but future work is planned on an analysis/synthesis system to estimate the control parameters automatically.

4. SPECTRAL MATCHING STRATEGY

The filter boost ranges of the five formant filters generally overlap to some degree so it is typically necessary to adjust the filter parameters to obtain the desired spectral contour. A current area of investigation is the development of a method to obtain the optimum filter parameters for each formant directly from an analysis of a musical waveform. Procedures for spectral estimation using all-pole or autoregressive (AR) models are well known [11], but the mathematical functions describing the proposed bandpass equalizer structure result in a set of non-linear equations. A computationally efficient scheme for performing the matching and non-linear equation solution is needed.

One possible approach is first to develop a conventional all-pole model for the desired waveform, then to segment the model into formant regions, and finally to match the model parameters of each formant individually with the bandpass equalizers. Another avenue for investigation involves the use of pre-calculated coefficients to match the analyzed spectrum to the closest stored spectrum, *à la* vector quantization.

5. CONCLUSIONS

In this paper a versatile subtractive synthesis system implemented with tunable bandpass equalizer filters has been presented. The conventional source-filter model is employed to obtain a variety of synthesis and cross-synthesis effects. Applications in singing synthesis are the primary motivation for this work, although a wide range of creative synthesis options can be addressed within the same framework.

6. ACKNOWLEDGMENTS

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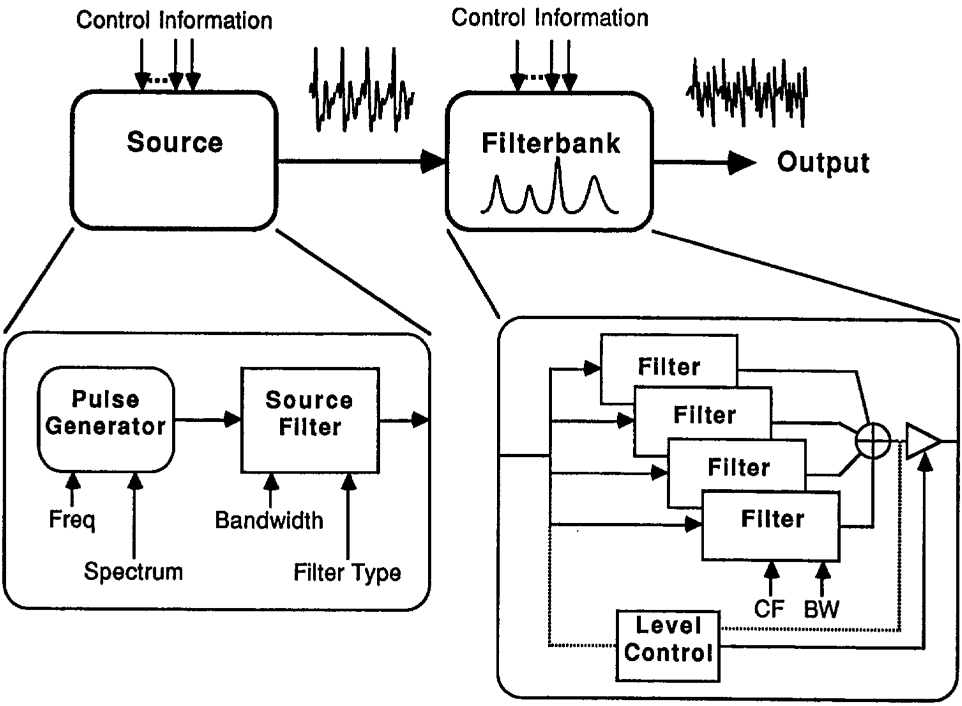

Figure 1: A block diagram of the conventional subtractive synthesis procedure.

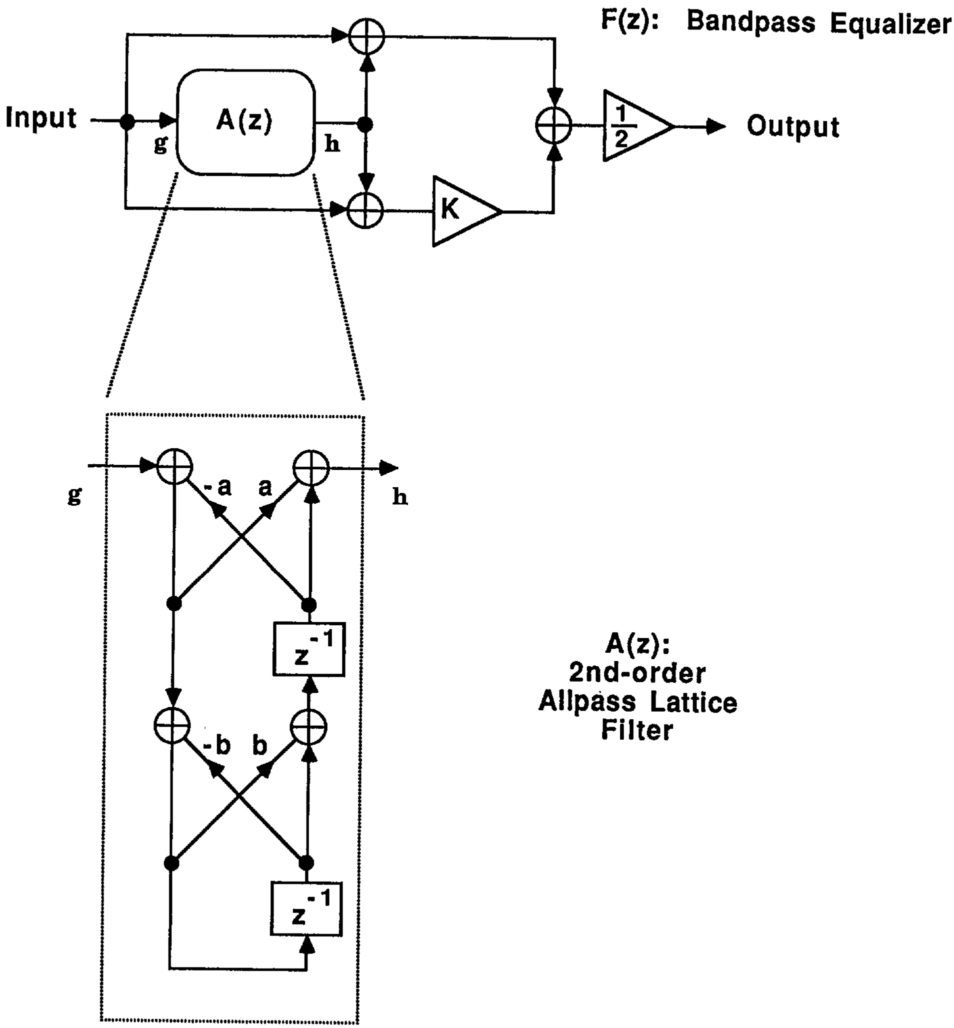

Figure 2: Structure of Regalia and Mitra 2nd-order bandpass equalizer filters. The multipliers a, b, and K are the filter parameters.

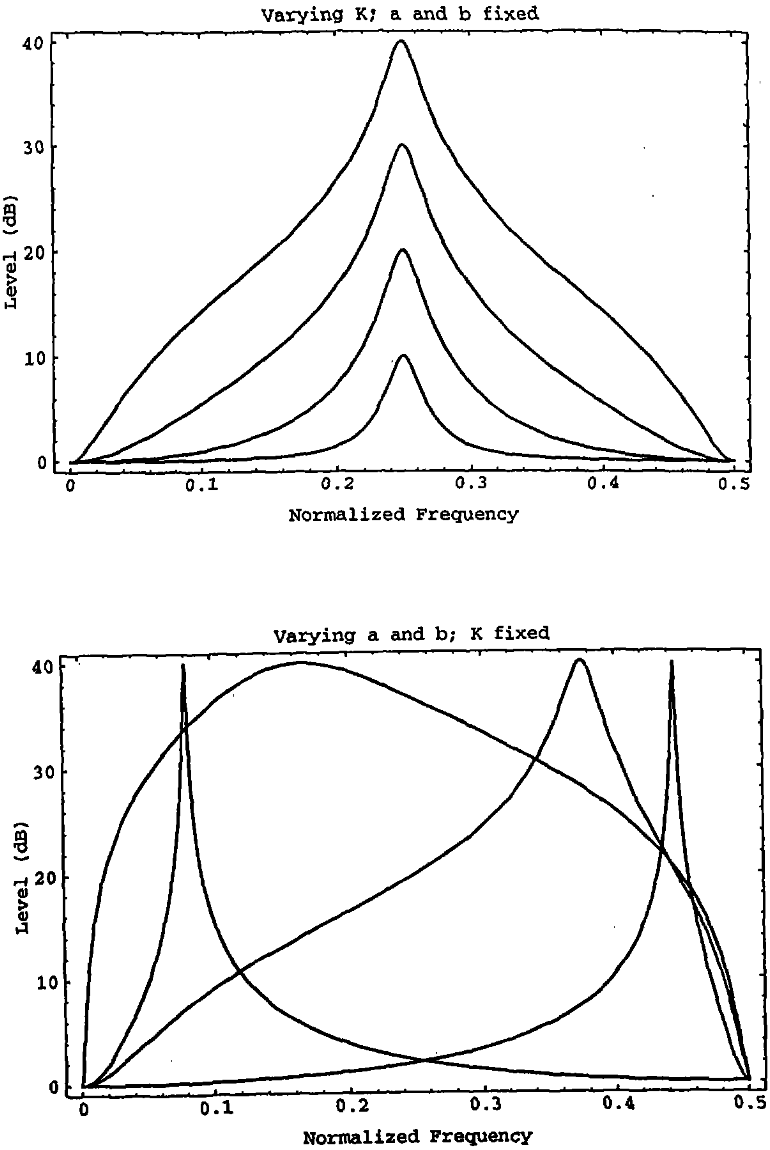

Figure 3: Filter magnitude response for various values of *a*, *b*, and *K*.

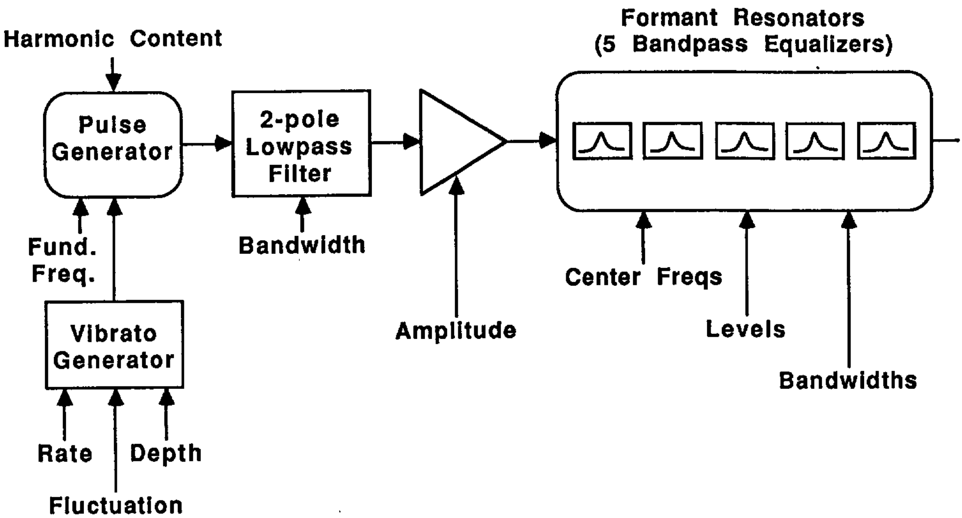

Figure 4: Standard elementary subtractive synthesis model for the singing voice.

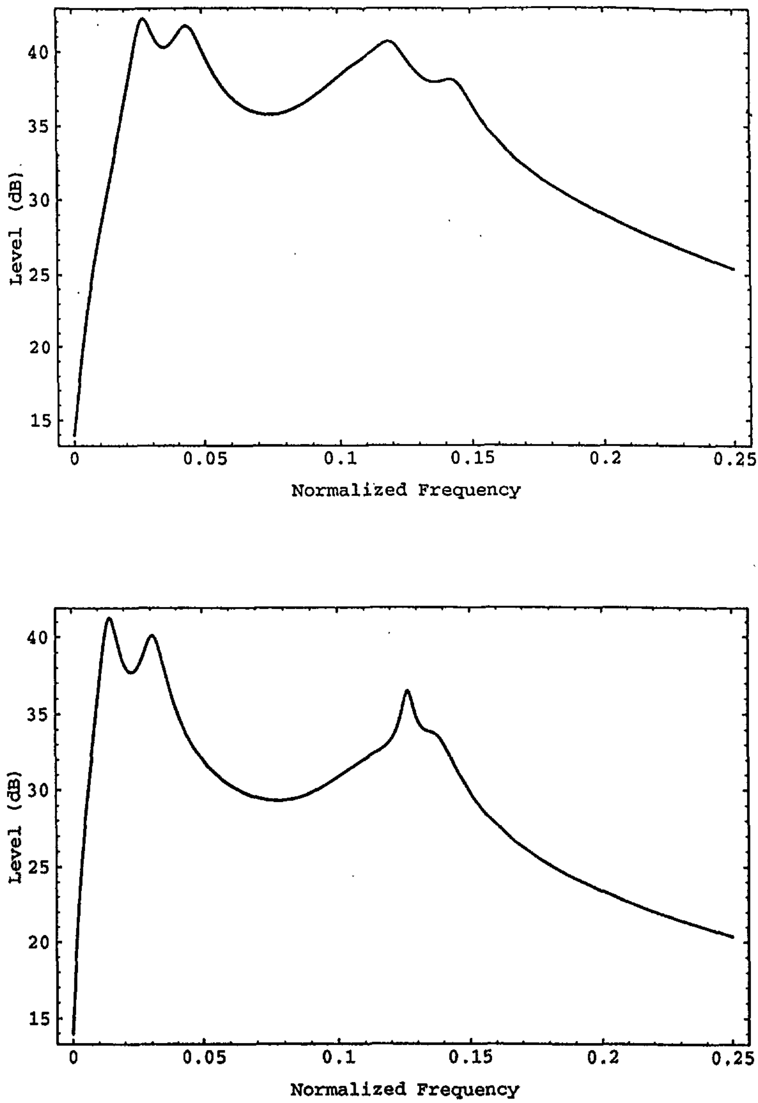

Figure 5: Filter spectra simulating several different vowel sounds.