1 BERLIN 33
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Mr. Yasunori Mochida, Director
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Dear Mr. Mochida:

First, I want to apologize for the delay in responding to your request for evaluation of the prototype. In returning to Berlin I caught the flu and have been, therefore, 'inactive'.

I will first make some general observations regarding the sound examples, second, specific suggestions, and third, other uses of the prototype circuit and possible modifications. Our research team at Stanford was very impressed with the initial implementation which you have achieved, confirming our beleief that digital processing is the future for electronic music and instruments.

I. General Observations:

The spurious frequencies which occur in the higher octaves of the sound examples result from fold-over around the 1/2 sample rate. Digital signal generation depends on the desired wave being band-limited, that is, having no frequency components greater than the ½ sampling rate. In order to avoid 'foldover' one must either increase the sampling rate (which is not efficient beyond a certain amount) or bandlimit the signal. In the case of FM the bandwidth of the signal is related more or less directly to the modulation index in the following way: the side frequency number n (of $J_n(I)$) is down at least 40 dB for $I \geq 2$ and $n=I+3$. (This approximation can be seen represented by contour line A in Fig. 8 of the AES paper.) Therefore, for a modulation index I=4, one can assume that any side-frequency of order 7 or greater will will be at least 40dB down. I f the carrier frequency is 1000 Hz and the index is 4, then we should expect frequencies of significance through 6000 Hz. In the case of a sampling rate of 10000/sec foldover would occur. (I am assuming the modulating frequency to also be 1000 Hz.) If the index were reduced to 2, however, all frequencies above ½ sampling rate would be insignificant. The generalization is

Limit = c + (I+3) * m, where c= carrier and m= modulating frequencies.

On the basis of the sampling rate, then, the index function should be scaled down as significant spectral components approach the ½ sampling rate, either by computation or by table look-up.

II. Specific remarks:

In tone #0, drum-like, you have succeeded in achieving a tone quality which we at Stanford had not yet attempted. This sequence in the upper octaves, except for the highest, sounds remarkably like a marimba or xylophone, congratulations! In order to preserve timbral continuity through the octaves, I believe that it will be necessary to change some of the parameters according to range.

To preserve the xylophone-like tone in the upper octave I suggest shortening the run-down time to one half or more, TR = 150 MS, as with this instrument the resonance decreases significantly with the smaller bars. To preserve the quality in the lower octave, I suggest decreasing the attack-time and increasing the index, for example, TA= 10-15 MS, TR=300 MS, and I1=20 to increase peak value of index from ca 2.5 to 5.

Tone #1 - (brass-like) Our experience has been that the low brass tones (tuba) are achieved simply as a function of range. That is, by simply maintaining the same parameters in the low registers will produce convincing low-brass tones. I have found that simulation is improved somewhat in the low registers when the attack is a bit longer, for example 100-120 MS in stead of 80 MS. The tone can be made a bit brighter by increasing the peak value of I1. The range we have used for brass tones is ca 4-6 with an average at about 5, for the steady state! - That is, the overshoot during the attack would exceed these values.

Introducing a Tremolo- A tremulant can be introduced into the brass tone (or any tone where KC/K1 is a simple ratio, such as basoon, clar, string etc.) in two ways:

1. By introducing a constant frequency difference in the modulating frequency. For example, if the carrier freq. is 440 Hz and the modulating freq. is 442 Hz the difference between the upper and the lower reflected side frequencies will produce a beat. I emphasize that the difference should be constant, for example, the octave above would be 880 and 882 for the KC and K1 respectively. Our experience has shown that if this difference is chosen quasi-randomly between say 1.5 and 2.5 Hz for each tone, the interest of tones in series is increased.

2. Use of secondary modulating wave and index. If K2 is made a constant instead of a ratio a similar effect can be achieved. For example, if in the case of the brass-tone, K2 is made a costant of K2= 2 Hz and the index is as shown by my addition on Tone #1 function plot (enclosed), additional side frequencies are produced around each of the harmonies at such a close interval that they are heard as beats. (In this application the ratio KC/K1 is unity, 1/1, without any difference.) This is explained by the fact that in FM, a carrier modulated by two sinusoids has frequencies according to the relatioship,

$$KC \pm nK1 \pm mK2 \quad \text{where n and m are integers}, 0,1,2,3\ldots.$$

This means of introducing additional frequency components has enormous potential which we have only begun to explore.

Tone #2 - Based on the premise that in string tones the bandwidth is both greater and more complex during the attack portion, I suggest beginning experiments with the functions as I have shown on Tone #2 function plot. Try also, a value for K2 of ca 1.64 in addition to 3.14 which you have used. In either case, the effect of the secondary modulating wave is to introduce inharmonicity during the attack. The exact amount can be controlled by either increasing or decreasing the peak value of I2 (here it is about 1, which I believe to be enough to produce a just noticeable 'scratch' or 'grit' during the attack. A tremolo can be added by technique #1 above.

Tone #3 - Intoduce tremolo effect #1 or 2 as indicated for Brass-tones. I should point out that both of these tremolo effects, in particular #2, give a broadness or resonant quality to the tone in addition to the tremolo. You might try other~~e~~ ratios of KC/K1, for example 4/1, 3/1.

Tone #4 - We will send you information from Stanford as soon as we have some results. I also believe that very convincing harpsichord tones can be achieved using the FM technique.

Tone #5 - For a plucked bass tone see the Tone #5 funtion plot. Note the amp increases linearly in 40-60 MS while the I1 decreases from ca 5 to 1. The decay should follow immediately or with a very short steady state (no more than 100 MS). I have only tried these tones in the octave C2 or two octaves below middle C. For the upper octaves I suspect that the attack time will have to be shortened.

Tone # 6 & 0 - For Bell-like the 0 function plots are more correct. There are a large number of possibilities here depending on the ratio KC/K1 and the peak index I1. For example, the principle interval above the carrier frequency can be determined in the following way: if the interval of a 4th is desired between the carrier and first significant side frequency, the ratio of a 4th = 4/3, produced by a ratio of c/m = 3/7, since the first lower side frequency is determined by c-m or 3-7 = -4 = +4 or KC/K1 = 1/2.33. Similarly, a major 3rd = ~~5/3~~ 5/4, c/m = 4/9, KC/K1=1/2.25. A major 6th = 5/3, c/m = 3/8, KC/K1=1/2.66. The rule of summing the numerator and ~~denomerate~~ denominator of the interval ratio for the modulating freq. is obvious. By using peak index values for I1 which are not too great, as small as even 3-4, one can simulate chimes as well as bells. It is important the index always approach zero as you have shown it on the second bell-like function plot (2nd Tone #0).

III. Other Uses and Modifications

The phase angle of the carrier and modulating waves does change to some extent the energy distribution in the spectrum. Our preliminary investigations show that the change is noticable but not striking. I will ask our digital signal processing expert at Stanford, A. A. Moorer to send you that information, as it demanding of more complex mathematics than I can manage.

The use of the secondary modulating wave as a means of introducing either a tremulant or inharmonicity can be enormously useful. Even wind instruments, clar and basoon, might be made more realistic by adding a small amount of inharmonicity during the attack.

An important consideration in realistic simulation of instrument tones is that the spectral composition of a tone may change radically according to the range (the continuity of brass tones through a number of registers with one set of paramenters may be an exception). It would be useful, therefore, to structure the simulation algorithms around tables from which parametric values, for example peak index values, are selected according to range. When I return to Stanford in April we can do some research to determine what these tables might look like.

The use of other waves in the table other than sine is perfectly predictable, but very complex acoustically. One can see that with the addition of only one frequency as with the secondary sinusoidal modulating wave the spectrum becomes enormously complex. I have to date, then, restricted my research to the simpler cases in order to exploit the possibilities to the fullest. I do not mean to imply that there are no useful results in using complex waves, and we certainly should investigate the possibilities. The spectrum would be composed of the sum of frequencies in the relation

$$KC \pm lK1 \pm mK2 \pm nK3 \ldots \quad \text{where } l,m,n.. \text{ are } 0,1,2,3\ldots$$

with amplitudes multiplicative $J_o(I)*J_l(I1)*J_m(I2)*\ldots,$

where for every harmonic in the modulating wave there is a term Kn. A sawtooth wave will produce an enormous complexity therefore.

If it is of interest to you now to introduce a vibrato (sub-sonic frequency modulation) in the prototype please let me know and I will send you the necessary information. A very interesting alternative is to sinusoidally modulate the index over ~~and~~ a range of ca + and - 25% at a rate of ca 5-6 Hz. In at least some cases this amplitude modulation is indistinguishable from vibrato. Note that each freq. component would increase or decrease in amplitude independently according to the slope of the Bessel function controlling its amplitude.

I hope that this will prove helpful and ask you, please, to contact me should there be any results or descriptions which are unclear. We will also get any information regarding piano and harpsichord tones to you as soon as the research is complete.

Sincerely,

John M. Chowning