Application Note AN3201-05: Reverberation Modeling an Auditorium
by Chris Maple

Introduction

There are several approaches to designing a reverb algorithm, of which three are predominant. The first – illustrated in this application note – is to model a physical space, whether it corresponds to a real space or not. Another is to create a complex digital filter using plenty of delay, adjusting it to create the desired sonic characteristics without regard to a physical model. A third approach is a blend of the first two, modeling reality to some extent but not requiring the resulting algorithm to emulate a defined physical space (see Application Note AN3201-04).

Modeling a physical room has value in that it can help design a room that has good acoustic properties for a particular application, or that it can help produce the audio image of a particular room from a closely microphoned recording. Difficulties include accurately modeling the shape and materials of the room, and sound travel paths even if those are known. This is a non-trivial set of calculations, and the DRE only has the capability to model a small part of what will go on in any complex acoustic environment.

What Makes A Good Reverb?

Historically, reverb has been used to increase the size of an audience that instruments or a human voice can reach, by reinforcing the original sound. With electronic sound reinforcement available, reverb is mostly used for esthetic appeal, based somewhat upon what we're used to and what does not destroy the intelligibility of the material.

Some rules are available based upon a great deal of testing that's been done by many investigators over the years. For instance, "Beneficial early reflections arrive within about 20ms of direct-sound arrival. A concentrated echo more that 50ms late is a serious acoustical defect." (Electronic Engineers' Handbook by Fink & Christiansen) Disobeying that rule turns a rimshot or a castanet into an annoying clatter. Generally, the 60dB decay time should be at least 0.7 seconds, and for very large rooms the decay time at low frequencies can be a little more than 2 seconds but less than 1.4 seconds above 1kHz.

Algorithm

Room Dimensions
I have assumed a rectangular auditorium 60' wide with a 40' ceiling. The listener is 75' from the back wall and 50' from the stage, centered in the room. The 40' stage is centered and sticks out 3' into the room. The stage is 4' high, level with the listener's ears. The performers' audio comes from a point 5' above the stage, 8' back from the edge. There are two performers, separated by 16'. The stage is 22' deep; at the back of the stage 3' from the wall is a curtain.

Reflectivity
The reflectivity of the walls is chosen to attain a 2 second decay time. The side walls have a reflectivity of 0.82. The stage takes half the area of the front wall, returning 85% of incident sound while the wall around it returns 90%. The curtain reduces reflections off the back wall so that only 85% of the incident sound is returned below 630Hz. All sound into the wings are totally absorbed. The auditorium back wall reflects 64%. Additionally, walls are not perfectly smooth; there are doorways and decorations and beams, etc. These are modeled with a crude IIR filter, which reduces high frequency reflections. The ceiling reflects 80%, and the stage floor reflects 90%.

Reflection Paths
The listener's ears are 58.76' from the performer's mouth, and this is the baseline dimension. Assuming that sound radiates equally well directly backwards (not true), the path from his mouth, through the backstage curtain, off the wall, through the curtain again and out to the listener's ears is 86.52', or 27.76' further. 27.76' is 1182 samples at 48 kHz. The sound is attenuated by the ratio of the distances, the passages through the curtain, and the reflection off the wall. Other sound paths are calculated in a similar manner. Some sound paths are so close to straight-ahead, or so diffuse, that they are assumed monaural, others are treated as stereo.

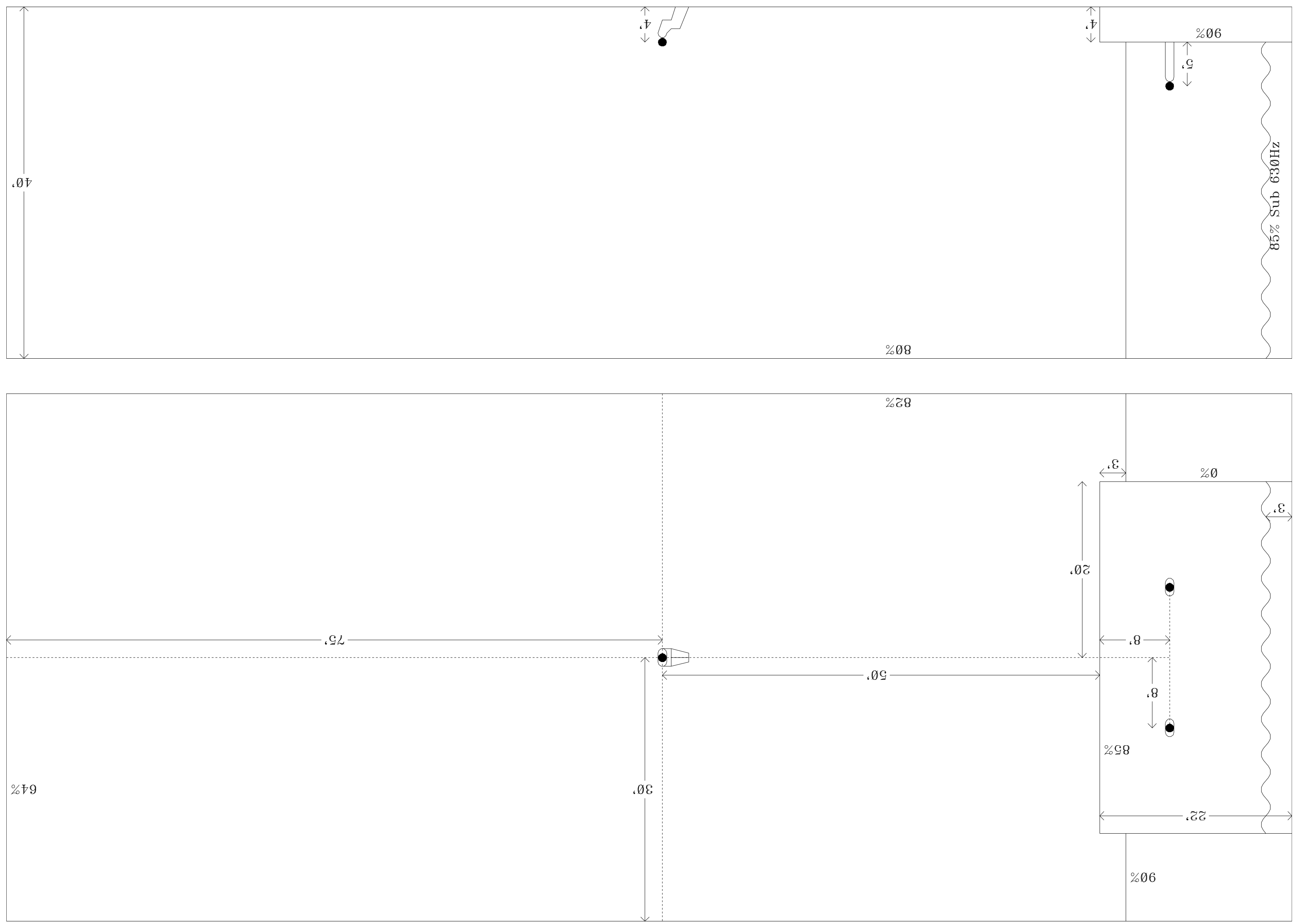
I distinguish (mostly early) reflections from the reverberant field. The latter I consider to be reflections which repeatedly travel between parallel surfaces, i.e. the front and back walls, the two side walls. I assume that the seats and audience absorb everything that hits them, thus there is no floor-ceiling reverberant mode. A good question is: "What excites the reverberant modes?" Properly, the sound source for the reverberation should be coming from the stage, i.e. the performers and reflections off the backstage wall and the stage floor. However, the sound at the listener's ears from direct and early reflections is already premixed, a convenience not to be overlooked when the number of available instructions is limited.

Algorithm Adjustments
Complex reverb algorithms can do what a real room does: provide deep notches and high peaks in the audio response. When the DRE gets a large signal and models a peak, it will do what a room does not: it will clip and thus distort. To prevent this, if the audio characteristics are to remain unchanged, either the amplitude of the input signal must be reduced or the gain through the DRE must be reduced. It is possible for intermediate points in the algorithm to clip even if the output doesn't clip, so look out for this. In this application note the gain through the DRE has been greatly reduced to prevent clipping at peaks. Still, a maximum amplitude input at 410 Hz will clip.

To reduce the headphone effect of the soundstage appearing inside the head, some head modeling has been done. Sounds approaching the right ear from 50 or 40 degrees off center are delayed 0.4ms and 0.32ms respectively, and pass through a 1.5kHz, 12dB lowpass shelving filter before being applied to the left ear. Figures for 10 degrees are 0.08ms, 1.5kHz, and 6dB. This is not a complex enough head model to be completely successful.

If you find that the sound of your reverb is muddy or annoying, the easiest change to make is to increase the ratio of dry (input) signal to reflected and reverberant signal. It may then no longer accurately model the room in question, but you want it to sound good, right?

Conclusion
After all this explanation and the math involved, the result doesn't sound as good as the reverb in AN3201-04: some directional clues are lost, and the strength and duration of the reverb makes fast-paced music somewhat disorganized. The spoken voice can become unintelligible. Much symphonic music, however, sounds only moderately different from the unaltered signal and the sensation of being in the audience in a big auditorium is achieved. The deficiencies may mean that the modeling of the auditorium isn't good enough (likely) or that such an auditorium wouldn't sound very good anyway (also likely). There are, after all, performance environments that are not acceptable for certain types of music. Certainly, it is easy to make a reverb that sounds poor, even if it models a real room. In any case, these application notes can be a starting point for designing reverb algorithms that fit your requirements.


40'
4'
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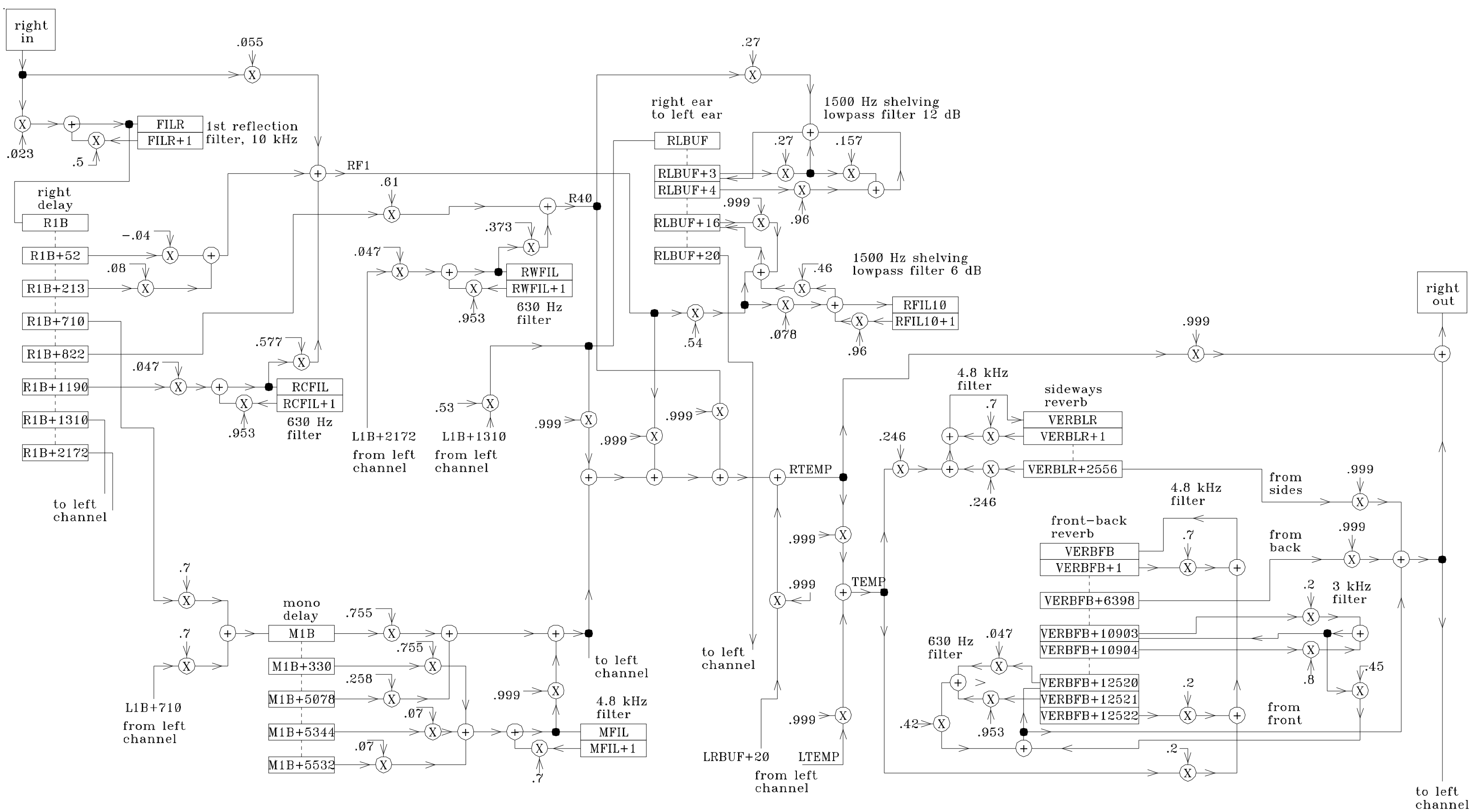
85%
64%
30'
22'
90%



Graphical representation of the processing for the right channel and the monaural components shared by left and right.

Source code

```
; File: AN3201-05.asm
; Auditorium reverb
; Author: Chris Maple

MEM    FILR   2          ;lopass IIR filter (-6dB 10 kHz) from right performer
MEM    FILL   2          ;from left performer
MEM    R1B    2173       ;single reflection buffer from right performer
MEM    L1B    2173       ;single reflection buffer from left performer
MEM    RF1    1          ;right front
MEM    LF1    1          ;left front
MEM    L40    1          ;left wall reflections to left ear at 40 degrees from front
MEM    R40    1          ;right wall reflections to right ear at 40 degrees
MEM    RL1    1          ;right performer reflections off left wall
MEM    LR1    1          ;left performer reflections off right wall
MEM    LRBUF  20         ;sound going from left ear to right ear
MEM    RLBUF  20
MEM    LEFIL  2          ;IIR filter for previous
MEM    REFIL  2
MEM    RFIL10 2          ;IIR shelving filter, -6dB above 1500 Hz; 10 degree path
MEM    LFIL10 2
MEM    LWFIL  2          ;1st refl 630 Hz backstage-to-left filter for right performer
MEM    RWFIL  2          ;1st refl 630 Hz backstage-to-right filter for left performer
MEM    MFIL   2          ;summed signals 2nd reflection mono filter
MEM    RCFIL  2          ;rear-stage-wall-curtain filter, right performer, 630 Hz
MEM    LCFIL  2          ;rear-stage-wall-curtain filter, left performer, 630 Hz
MEM    LTEMP  1
MEM    RTEMP  1
MEM    TEMP   1
MEM    M1B    5533       ;monaural reflection buffer
MEM    VERBLR 2556       ;side-to-side reverberation buffer
MEM    VERBFB 12522      ;back to front reverberation buffer
MEM    REFRESH 1024      ;code uses some dummy writes, does refresh at same time

;---------------------
RZP    ADCR   K=.023     ;right performer source times .023
RAP    FILR+1 K=.5         ;1 reflection filter, LF amplitude = .046 x input
WAP    FILR   K=0          ;blur for surface roughness, -6dB point is 10.3 kHz
WZP    R1B    K=0        ;start of right reflection buffer
RZP    R1B+1190 K=.047   ;630 Hz filter for backstage reflection through curtain
RAP    RCFIL+1 K=.953    ;filter's feedback term        distance is +27.74'
WZP    RCFIL  K=.577       ;save filter output, scale for distance & relectivity
RAP    R1B+52 K=-.04     ;diffraction/impedance mismatch at stage edge, +1.21'
RAP    R1B+213 K=.08     ;reflection off clutter at right performer's feet, +5.0'
RAP    ADCR   K=.055     ;direct
WZP    RF1    K=0        ;save direct + single reflections from front (10 degree)
RZP    L1B+2172 K=.047   ;left performer, stage rear wall and right wall, +50.99'
RAP    RWFIL+1 K=.953      ;This filter is modeling the stage rear wall
WZP    RWFIL  K=.373       ;Save the filter output, scale for distance & reflectivity
RAP    R1B+822 K=.61     ;reflection off right wall, +19.3'
WZP    R40    K=0        ;save reflections off right wall entering right ear at 40 deg
RZP    L1B+1310 K=.53    ;left performer reflection off right wall, +30.76'
WZP    RLBUF  K=0        ;start the right ear to left ear buffer
```

```
;------ repeat the above, swapping left and right
RZP    ADCL   K=.023
RAP    FILL+1 K=.5
WAP    FILL   K=0
WZP    L1B    K=0
RZP    L1B+1190 K=.047
RAP    LCFIL+1 K=.953
WZP    LCFIL  K=.577
RZP    L1B+52 K=-.04
RAP    L1B+213 K=.08
RAP    ADCL   K=.055
WZP    LF1    K=0
RZP    R1B+2172 K=.047
RAP    LWFIL+1 K=.953
WZP    LWFIL  K=.373
RAP    L1B+822 K=.61
WZP    L40    K=0
RZP    R1B+1310 K=.53
WZP    LRBUF  K=0

;-------- process the rest of the right ear to left ear buffer
RZP    R40    K=.27         ;40 degree path
RAPC   RLBUF+3 K=.27        ;+ 50 degree path, save result in C
WZP    REFRESH K=.157       ;scale product to .042=.27*.157, dummy write does refresh
RAP    RLBUF+4 K=.96        ;filter feedback term (LP shelf 1500 Hz, HF -12dB)
WCP    RLBUF+3 K=.73        ;save new feedback term; create filter output
WZP    RLBUF+4 K=0          ;save output at +0.08ms
RZPC   RF1    K=.54           ;10 degree path from right performer
WZP    REFRESH+0x40 K=.078    ;save .54 product as refresh, make new .042 product
RAP    RFIL10+1 K=.96       ;filter feedback term
WCP    RFIL10 K=.46           ;save new feedback term and create filter output
RAP    RLBUF+16 K=.999      ;add filtered 10 degree path at +0.32ms
WZP    RLBUF+16 K=0         ;save back into head delay line

;-------- left ear to right ear buffer
RZP    L40    K=.27
RAPC   LRBUF+3       K=.27
WZP    REFRESH+0x80 K=.157
RAP    LRBUF+4       K=.96
WCP    LRBUF+3       K=.73
WZP    LRBUF+4       K=0
RZPC   LF1    K=.54
WZP    REFRESH+0xC0 K=.078
RAP    LFIL10+1 K=.96
WCP    LFIL10 K=.46
RAP    LRBUF+16 K=.999
WZP    LRBUF+16 K=0

;------- All other reflections are considered to be neither from the left nor from the
;------- right, and are treated as monaural. The mono reflections start at 88.98', or
;------- 30.22' more than the direct path. Sum of left and right, times .7; later
;------- coefficients are x1.43
RZP    R1B+710 K=.7         ;right channel +30.22' mark
RAP    L1B+710 K=.7         ;left channel
WZP    M1B    K=.755        ;start the mono buffer and scale the ceiling reflection
RAPC   M1B+5078 K=.258      ;208.21', 119.23' into mono delay, off auditorium back wall
           ;The remaining reflections are double reflections and need to be blurred.
RZP    M1B+330 K=.272       ;7.75' into mono delay, stage floor and auditorium ceiling
RAP    M1B+5344 K=.07       ;+125.48' delay, same side wall as performer & aud. back wall
RAP    M1B+5532 K=.07       ;+129.91' delay, opposite side wall & auditorium back wall
RAP    MFIL+1 K=.7          ;4.8kHz filter feedback term
WCPC   MFIL   K=.999        ;save previous result in filter, add C, put sum in C
```

```
;--------
; sum all right ear audio except reverberant field
RCP    RF1    K=.999        ;right performer front reflections and direct
RAP    R40    K=.999        ;40 degree inputs
RAP    RLBUF  K=.999        ;50 degree input
RAP    LRBUF' K=.999        ;right ear audio delayed relative to left ear
WZP    RTEMP  K=0           ;Save sum.
; sum all left ear audio except reverberant field
RCP    LF1    K=.999
RAP    L40    K=.999
RAP    LRBUF  K=.999
RAP    RLBUF' K=.999        ;sum of left-ear non-reverberant audio
WZP    LTEMP  K=.999

;--------
; Do the auditorium reverberant modes. Since the listener is assumed to be exactly
; centered in the 60' width, the width may be considered as 30' with one perfectly
; reflecting wall. The length of the auditorium is considered to be evenly split
; between a section 128' long from the front to the back, and another section 147'
; long from the back of the auditorium to the rear of the stage. The floor-to-ceiling
; reverberant mode is assumed not to exist, as the seats and audience would absorb
; all reflections directed at the floor.

;------------ side-to-side reverberation
RAP    RTEMP  K=.999        ;add left and right temps as source for reverberant fields
WZP    TEMP   K=.246          ;save, start lopass with gain = .82, F6db = 4.8k
RAP    VERBLR' K=.246       ;other input to lopass is end of 60' delay
RAP    VERBLR+1 K=.7          ;this is the filter's feedback term
WZP    VERBLR K=0             ;save the filter output at the side reverb input

;------------ front-to-back reverberation
RZP    VERBFB' K=.2         ;One input for the backward-travelling delay is buffer end.
RAP    TEMP   K=.2          ;The other is TEMP. Same filter type as previous, this is for
RAP    VERBFB+1 K=.7        ;the reflection from the back of the auditorium.
WZP    VERBFB K=0           ;Save the filtered audio as the beginning of the delay buffer
RZP    VERBFB+10903 K=.2      ;Sound has passed listener going forward. This 3.0kHz
RAP    VERBFB+10904 K=.8      ;filter serves as the input to reflections off both the
WZPC   VERBFB+10903 K=.45     ;auditorium front wall (K=.45) and the backstage wall.
RZP    VERBFB'-2 K=.047         ;backstage wall with curtain needs 630 Hz filter
RAP    VERBFB'-1 K=.953         ;
WCP    REFRESH+0x100 K=.42      ;scale filter out, + front wall refl in C. Write is dummy
WAP    VERBFB'-2 K=0          ;Save the split delay sum, the backward-travelling wave.
RAP    VERBFB+6389 K=.999   ;Add the forward-travelling wave as it passes the listener.
RAPC   VERBLR' K=.999       ;Add the side-to-side reverberation, save total reverb in C.
RCP    RTEMP  K=.999        ;Add reverb to non-reverberant right
WAP    OUTR   K=0           ;Total right ear output
RCP    LTEMP  K=.999
WAP    OUTL   K=0           ;Total left ear output

;------------ refresh
RZP REFRESH+0x140
RZP REFRESH+0x180
RZP REFRESH+0x1C0
RZP REFRESH+0x200
RZP REFRESH+0x240
RZP REFRESH+0x280
RZP REFRESH+0x2C0
RZP REFRESH+0x300
RZP REFRESH+0x340
RZP REFRESH+0x380
RZP REFRESH+0x3C0
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NOTICE

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Application note revised March, 2005