HOW SMOOTH DO YOU THINK I AM: AN ANALYSIS ON THE FREQUENCY-DEPENDENT TEMPORAL ROUGHNESS OF VELVET-NOISE

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ABSTRACT

Velvet noise is a sparse pseudo-random signal, with applications in late reverberation modeling, decorrelation, speech generation, and extending signals. The temporal roughness of broadband velvet noise has been studied earlier. However, the frequency-dependency of the temporal roughness has little previous research. This paper explores which combinative qualities such as pulse density, filter type, and filter shape contribute to frequency-dependent temporal roughness. An adaptive perceptual test was conducted to find minimal densities of smooth noise at octave bands as well as corresponding lowpass bands. The results showed that the cutoff frequency of a lowpass filter as well as the center frequency of an octave filter is correlated with the perceived minimal density of smooth noise. When the lowpass filter with the lowest cutoff frequency, 125 Hz, was applied, the filtered velvet noise sounded smooth at an average of 725 pulses/s and an average of 401 pulses/s for octave filtered noise at a center frequency of 125 Hz. For the broadband velvet noise, the minimal density of smoothness was found to be at an average of 1554 pulses/s. The results of this paper are applicable in designing velvet-noise-based artificial reverberation with minimal pulse density.

1. INTRODUCTION

Velvet noise is a sparse pseudo-random noise sequence, which consists of ternary values (-1, 0, and 1) [1] and has a constant power spectrum [2]. Velvet noise was originally proposed by Karjalainen and Järveläinen to model room reverberation [1]. It is known that late reverberation resembles exponentially-decaying, filtered white noise [3, 4, 5]. Broadband velvet noise has been shown to retain its perceived smoothness with lower pulse densities in comparison to other types of sparse noise sequences [6]. At 2000 impulses per second, velvet noise has been shown to sound smoother than Gaussian white noise (GWN) [6].

The perceived temporal smoothness of velvet noise has been investigated mostly on broadband noise sequences [6]. Karjalainen and Järveläinen [1] made an initial study on the frequency-dependency of the temporal roughness, where lowpass filtered velvet-noise, with a cutoff frequency of 1.5 kHz, was shown to sound smoother than GWN with a pulse density of 600 pulses/s. This paper investigates further the frequency-dependent psychoacoustic temporal roughness of velvet-noise sequences. Having a clear understanding of the frequency-dependency of the temporal roughness can help in the design and optimization of reverberation models based on sparse noise sequences.

The earliest sparse-noise-based reverberation algorithm was proposed by Rubak and Johansen [7, 8]. Their algorithm is based on totally random noise (TRN), which is a type of sparse pseudo-random noise with an equal probability of any sample having a non-zero value. The proposed minimal pulse density for producing high-quality noise with the TRN was reported to be between $2000 - 4200$ pulses/s, however for a lowpass filtered noise with cutoff at 8 kHz.

Rubak and Johansen also proposed a recursive structure for computational efficiency [7], which was further improved by Karjalainen and Järveläinen [1] by replacing the TRN with velvet noise and by introducing time-variation. The time variation was introduced to reduce the periodicity of repeating the same short velvet-noise sequence inside the recursive structure. A further problem arises from the time-variability which creates warbling especially on stationary input sounds [1, 9, 10]. An alternative solution to mitigate the periodicity problem was proposed in [10], where interleaved velvet-noise sequences hide the repetitiveness.

A different approach to reverberation modeling was taken in [4, 5], where filtered velvet noise segments were concatenated to model target late-reverberation. It was shown empirically that lower pulse density could be used towards the end of the model response, where the bandwidth was reduced. Recently, it was also shown that colored velvet noise can be generated directly by controlling the pulse location distribution [11, 2]. A practical algorithm for generating velvet noise with a lowpass spectrum, called dark velvet noise (DVN), was later proposed in [12]. The cutoff of DVN can be varied in time to generate characteristic late-reverberation, where the low frequencies decay slower than the high frequencies.

Another application for velvet-noise is to implement an efficient decorrelator [13, 14]. An optimization scheme for minimizing the spectral coloration introduced by the velvet-noise decorrelator was proposed in [14]. Velvet noise has been also used in hybrid reverb structures combining it with feedback delay networks (FDN) [15, 16]. Short velvet-noise filters are applied either within the feedback matrix [16] or at the inputs and outputs of the FDN [15]. Additionally, velvet-noise has been used in vocoder-based speech generation by serving as excitation signals [17].

In this paper, the perceptual temporal roughness of velvet noise at octave bands as well as at lowpass bandwidths with the octave band center frequencies as the cutoffs are studied in a perceptual test. Additionally, the time-domain smearing of various filter orders is investigated objectively to narrow down the filter

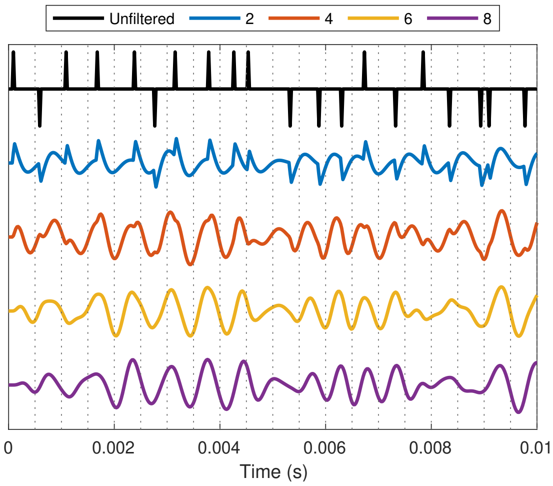


Figure 1: *Velvet-noise sequence, filtered with an octave-band filter at the center frequency $F_c = 1.5$ kHz, using different filter orders $N \in \{2, 4, 6, 8\}$ from top to bottom. The consecutive plots are offset for better visualization. The grid size $T_d = 0.5$ ms is shown with a dotted line.*

selection for the listening test. An adaptive perceptual test was implemented in Matlab, where the subjects control the density of test velvet noise signals. The test aims to find perceptual density thresholds where each test signal still sounds as smooth as a reference velvet noise signal with the pulse density of 2000 pulses/s.

The rest of this paper is organized as follows. Section 2 gives relevant background on velvet noise and temporal roughness. Section 3 describes the design of the filters for the listening test. Section 4 introduces the listening test setup. The results of the listening test are analyzed in Section 5. Section 6 concludes the paper and comments the future work.

2. BACKGROUND

This section explains the concepts behind velvet noise and temporal roughness.

2.1. Velvet Noise

Velvet noise is a sparse random noise sequence comprised of only sample values of -1, 0, and 1. Each frame contains a single randomly placed impulse with randomized sign; the rest are zeros, leading to a sparser noise than GWN. The number of nonzero impulses per second is defined as the pulse density ρ, i.e.,

$$\rho = \frac{f_s}{T_d}, \tag{1}$$

where T_d refers to the average distance between impulses measured in samples and f_s refers to the sampling rate. In this work, a sampling rate of $f_s = 44100$ Hz is used for all generated signals.

Karjalainen and Järveläinen [1] found that a pulse density of ρ = 1500 pulses/s satisfied the aim of minimal pulse density and maximal smoothness. This was the sweet spot for a perceived smoother noise than GWN. To prevent gaps and clusters of sample values which contribute to the perception of temporal roughness, the impulse locations are determined as [6]

$$k(m) = \lfloor mT_d + r_1(m)(T_d - 1)\rceil, \tag{2}$$

where m refers to the pulse counter while $r_1(m)$ refers to the sequence of uniformly distributed random values between 0 and 1, and $\lfloor\cdot\rceil$ is the rounding operation. The velvet-noise sequence is computed with

$$s(n) = \begin{cases} 2\lfloor r_2(m)\rceil - 1, & \text{when } n = k(m) \\ 0, & \text{otherwise,} \end{cases} \tag{3}$$

where n is the sample index and $r_2(m)$ is another uniform random number sequence to decide when an impulse will be 1 or -1. In Fig. 1, the black line at the top shows a broadband velvet noise sequence. Here, the impulses of 1 or -1 appear only once in each frame.

Additionally, velvet noise is featureless with a flat power spectrum [6]. The computing time of a convolution with velvet noise is much shorter than that of GWN because it mainly consists of zeros and where there are impulses in velvet noise, the ones and minus ones are easy to multiply by [4, 5].

2.2. Temporal Roughness of Sparse Noise

Temporal roughness is a psychoacoustic quality of sparse noise sequences where the lower the pulse density the rougher the signal is perceived [1, 6]. Temporal roughness has not been fully defined in the literature but it might be directly related to the roughness which is defined as the sensation caused by amplitude-modulated sine waves with modulation frequencies between $15 - 300$ Hz range. The sensation reaches its maximum around modulation frequency of 70 Hz [18].

Rubak and Johansen [7] observed that when the delay of a comb filter is increased above 25 ms one starts to perceive roughness in the sound and the perception changes from the coloration to the time-domain character. The value 25 ms corresponds to a frequency of 40 Hz for the pulses of the comb filter, which falls close to the modulation frequency causing maximal roughness sensation. Furthermore, it is reported that the modulation signal does not have to be periodic to cause the perception of roughness [18] The random assignment of pulses in sparse noise sequence can be interpreted as pseudo-random amplitude modulation [19].

Velvet noise has the ability to sound smoother than GWN, leading to the question of at which pulse densities is velvet-noise perceived as smooth versus perceived as temporally rough. In previous studies on sparse noise, optimal pulse densities for still maintaining perceived smoothness were researched in multiple ways such as first-order lowpass filtering reverberation tails and using totally random noise [7, 8] as well as using velvet-noise [1] which gave results of $2000 - 4200$ pulses/s [7, 8] and 1500 pulses/s [1] as optimal smoothness, respectively.

3. FILTER DESIGN

In order to investigate the frequency-dependent temporal roughness of velvet noise, filtering is to be applied to the white velvet-noise sequences. In this research, both lowpass and octave-band filtering is applied. In this section, the properties of various filter orders are investigated, to narrow down the filter parameters for the perceptual test.

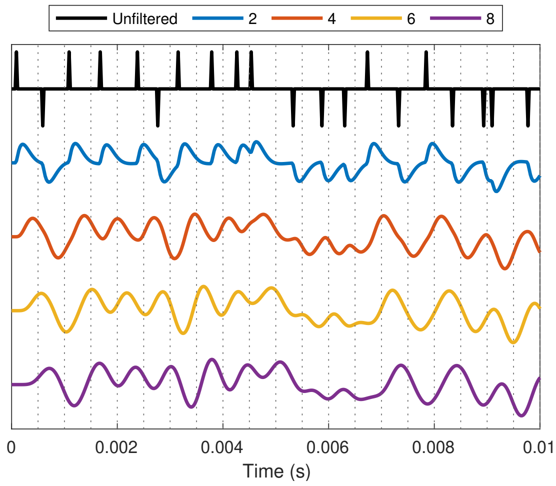
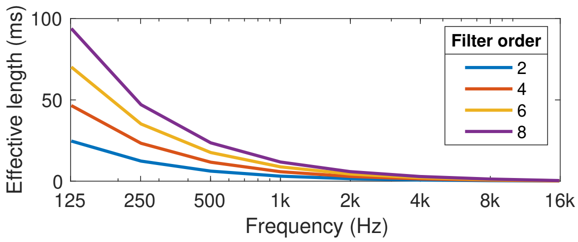
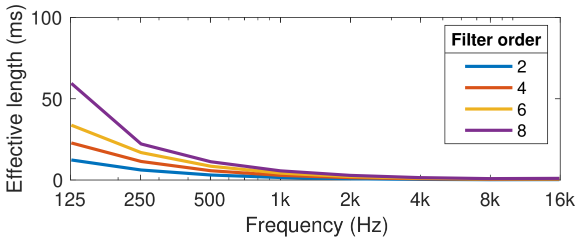


Figure 2: *Velvet-noise sequence, filtered with a Butterworth lowpass filter with the cutoff frequency $F_c = 1.5$ kHz, using different filter orders $N \in \{2, 4, 6, 8\}$, cf. Fig. 1.*



(a)



(b)

Figure 3: *Effective length of the (a) octave-band filters and (b) lowpass filters at the center frequencies $F_c \in \{125, 250, 500, 1k, 2k, 4k, 8k, 16k\}$ Hz, using different filter orders $N \in \{2, 4, 6, 8\}$.*

3.1. Time-domain Smearing

Time-domain smearing refers to the energy of a signal being spread across a longer period of time during playback which also means a loss of detail in the signal itself. Fig. 1 shows an example velvet-noise sequence (black) and its filtered versions (colored) with various filter orders of an octave band filter centered at $F_c = 1.5$ kHz. As the order of the filter increases the time response gets more and more smeared in time. With the second-order octave filter (blue) in Fig. 1, the pulse locations are still visible. A similar trend is shown in Fig. 2, which shows the velvet-noise sequence filtered with a lowpass Butterworth filter with various filter orders. The second-order lowpass filter shows more smearing than the second-order octave filter. This is why the fourth-order octave filters were used to compare against the lowpass filters.

Fig. 3a and Fig. 3b show the effective length of the octave-band filter and the Butterworth lowpass filter, respectively. Again, various filter orders are compared. The effective length is computed with Matlab function `impzlength` with a tolerance of -60 dB. Longer effective lengths of the filter will result in more time-domain smearing. The overall trend in Fig. 3 is that higher filter order and lower center frequency or cutoff frequency result in longer effective length. Furthermore, the difference in effective length between the lower and higher center frequencies grows with the filter order. Thus, for the listening test design we opted to use the second-order lowpass filters which introduce minimal smearing. For the octave filters order four was used, since the steepness of the second-order lowpass is most similar to that of a fourth-order octave filter.

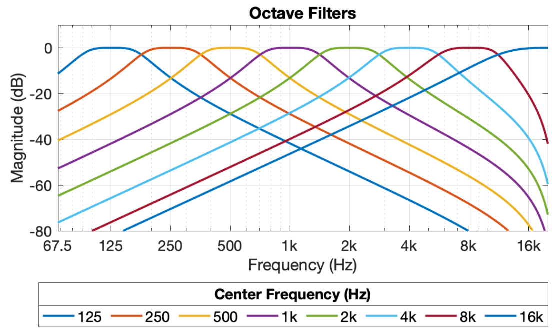
3.2. Listening Test Filters

Two types of filters were used in the final listening test: second-order Butterworth lowpass filters and fourth-order octave-band filters. A lowpass filter attenuates frequencies above a specified cutoff frequency and the frequencies below the cutoff are retained. The Butterworth lowpass filter [20] is often used in audio because of its maximally flat magnitude response in the passband and monotonic roll-off in the stopband. The magnitude responses of the used octave filters and lowpass filters are shown in Fig. 4a and Fig. 4b, respectively. Note: There is a slight shift of the lowpass passband versus the passband of the octave filters.

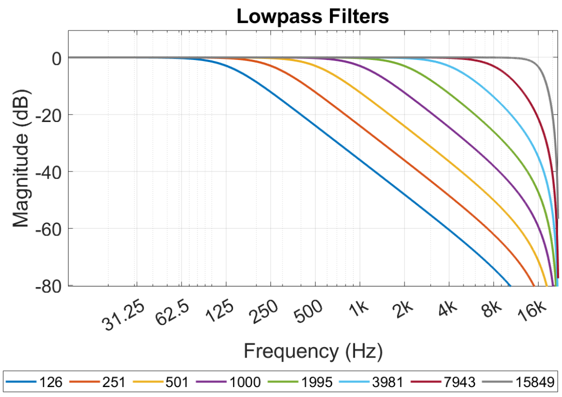
As for octave filters, an octave means an interval where there is a frequency ratio of 2:1, the upper frequency is twice the lower frequency. This is important when using the filter which consists of bandpass filters, the bandwidth of each filter will always be a 2:1 ratio. There are ten octave bands within the human hearing range. Octave filters are used because they can be utilized for measuring noise power at certain frequency ranges. They also give insight into the human hearing of temporal roughness which was the aim of this study. The magnitude response is shown in Fig. 4a. The center and cutoff frequencies were calculated using base 10. The nominal center frequencies of the octave filters used in the test are shown in Table 1 and is also shown by the peaks/centers of the octaves filter magnitude responses in Fig. 4a.

4. LISTENING TEST DESIGN

To test the perception of smoothness in velvet noise, a listening test was employed using Matlab App. As shown in Fig. 5 of the test user interface, there is a reference signal and a test signal. The reference signal was broadband velvet noise of 2000 pulses/s which has been shown to be perceived as smooth [1]. The participant was asked to find the lowest possible pulse density in which the test signal sounded as smooth as the reference. There was not a specific practice page, but testers were given a brief introduction to the sliders and signals where they could ask questions about the task. In this opportunity, they were able to familiarize themselves with the loudness level of the signals, the buttons and the slider



(a)



(b)

Figure 4: *(a) Fourth-order octave and (b) second-order Butterworth lowpass filter magnitude responses with center/cutoff frequencies one octave apart between 125 Hz and 16,000 Hz.*

function. The test required the participant to move the slider accordingly to match their perception of where the velvet noise signal is still smooth, but on the edge of being rough. The movement of the slider changed the pulse density of the velvet noise signal

Table 1: *Nominal center frequencies used to test the effect of octave and lowpass filters. For the lowpass filters, the center frequency was used as the cutoff frequency. Note: The lowest 2 octaves of the audio range are omitted, only octaves 3 to 10 were used in testing.*

Octave band	Nominal center frequency (Hz)
3	125
4	250
5	500
6	1000
7	2000
8	4000
9	8000
10	16000

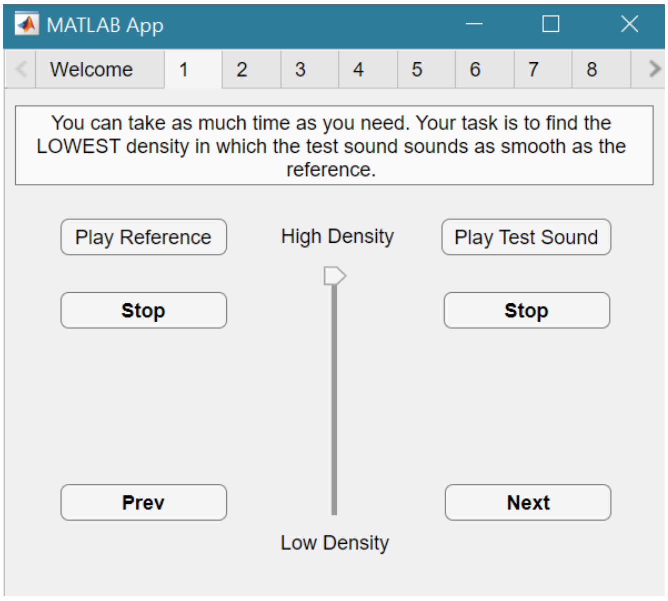


Figure 5: *User interface of the listening test. The participant chooses the lowest density where the test sound is still as smooth as the reference. The user can choose to play and stop the reference sound and the test sound. The participant can also go back to the previous or go to the next page using the respective buttons.*

and automatically played the test sound set to the chosen pulse density. Each page's velvet noise condition contained all possible pulse densities between 50 and 2000 and the reference remained at 2000 pulses per second for each test page. A pulse density of 2000 was chosen for the reference as previous papers on sparse and velvet noise respectively found already 1500 and 2000 as acceptably smooth pulse densities [1, 6].

The test signals presented were either unfiltered, filtered through an octave filter or through a lowpass Butterworth filter. Sound examples of the test signals are available on the companion web page of this paper[1] The filters used center frequencies or cutoff frequencies calculated from octave bands 3-10, shown in Table 1. The listening test was composed of two linked MATLAB apps, the first assessed octave filtering, while the second assessed lowpass filtered and broadband velvet noise. The center frequency change in the filters between pages was randomized. The lowest two center frequency bands were omitted due to being too low for users to perceive, and, for this reason, are not included in Table 1 either.

The loudness level for each signal was set based on the EBU R 128 standard at -23 LUFS (Loudness Unit Full Scale). LUFS is a loudness measurement based on the human perception of loudness. This was used over the counterpart of RMS (root-mean-square) power which is the average power of a signal without any weighting. Additionally, the use of LUFS was due to the frequency-dependent nature of loudness LUFS can account for. Normalizing the loudness between test signals was to ensure that rating was not influenced by how loud the signal sounded such as the perception that high frequencies are louder compared to low frequencies.

[1]`http://research.spa.aalto.fi/publications/papers/dafx23-vn-roughness`

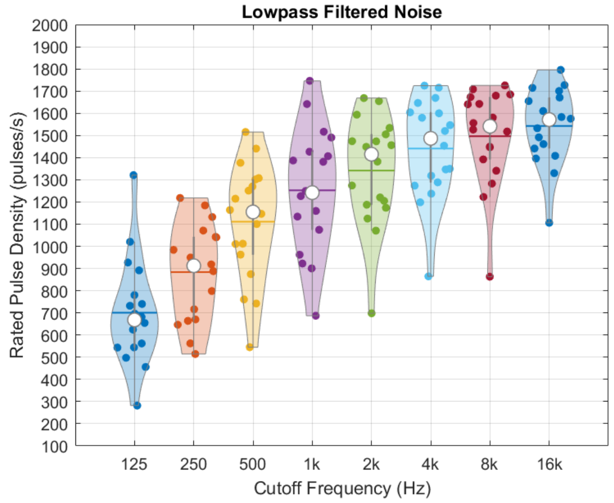


Figure 6: *Violin plot of minimal pulse density for smooth-sounding lowpass filtered velvet noise.*

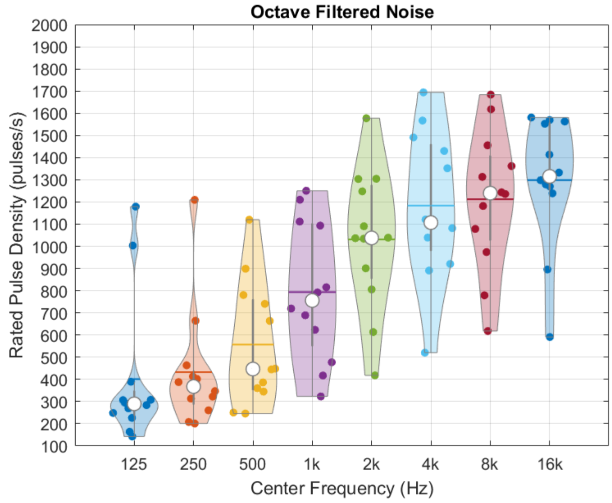


Figure 7: *Minimal pulse density for smooth-sounding octave filtered velvet noise.*

The final listening test sound pressure level of the isolated listening booths used for testing was calibrated to 60 dB [6] using a RA0045 G.R.A.S Ear Simulator. The simulator followed IEC 60318-4 regulations. Participants listened to the signals using Sennheiser HD-650 headphones.

There was a total of 34 listening test pages with 16 pages testing octave-filtered velvet noise, 16 testing pages for lowpass-filtered velvet noise, and two pages testing unfiltered velvet noise. This was done so that each signal of same conditions: filter type and center frequency or cutoff frequency, occurred twice during the test to compare individual differences and assess the reliability of the participants' ratings.

In total, there were 12 participants between ages 19 and 42 who completed the octave filtered velvet noise tests and 18 participants between ages 19 and 42 who completed the lowpass filtered velvet noise test all with previous experience in a formal listening test. For each test signal, there were two pages, and a correlation was calculated for each participant between the two pages for each center frequency/cutoff frequency. This was done using `corrcoef` in MATLAB. If the participant's mean correlation coefficient was below 0.5, meaning their answers were too different for the same stimuli to be considered, their data was discarded. Fortunately, no participants had a correlation coefficient under 0.8 and therefore, all participant data was assessed.

5. RESULTS

The results in Fig. 6 and Fig. 7 show the rated pulse density in the y-axis against the center frequencies of the 8 octave bands used in this study on the x-axis. The rated pulse density values come from the participant's task of setting the slider to where they perceive the lowest possible density where it still sounds as smooth as the reference signal. In both violin plots [21], the central white dot refers to the median of the data within the specific octave band center frequency. The means are indicated by a horizontal line in each violin. The median and mean values can be found in Tables 2 and 3. These values were calculated using an average of each participant's answers for each filter band and filter type respectively.The top and bottom of the thick grey line in the center of each plot refers to the first and third quartiles. Both plots seem to follow somewhat of a curve showing that there is some frequency-dependence on the perception of smoothness. Especially the octave-filtered velvet noise was rated with a consistently

Table 2: *Nominal cutoff frequencies used for lowpass cutoffs and their corresponding median and mean results.*

Cutoff frequency (Hz)	Median (pulses/s)	Mean (pulses/s)
125	668	725
250	912	896
500	1155	1120
1000	1242	1260
2000	1415	1350
4000	1487	1455
8000	1541	1509
16000	1571	1548
Broadband	1587	1554

Table 3: *Nominal center frequencies used for octave filters and their corresponding median and mean results.*

Center frequency (Hz)	Median (pulses/s)	Mean (pulses/s)
125	290	401
250	368	433
500	447	557
1000	756	794
2000	1038	1035
4000	1107	1183
8000	1240	1212
16000	1315	1299
Broadband	1587	1554

lower pulse density than for the low pass filtered velvet noise.

Fig. 6 demonstrates that lower cutoff frequencies allowed the pulse density of the velvet noise to be lower than that of broadband and octave-filtered signals. The signals were rated at nearly half the pulse density of broadband in the lowest octave band center frequency cutoff of 125 Hz, whereas in the highest human hearing octave band frequency cutoff, there was little difference.

We found the broadband velvet noise had a median of 1587 pulses/s and a mean of 1554 pulses/s where the signal still sounded as smooth as the reference signal, as shown in Fig. 6, which is similar to results in the original velvet noise study [1], where 1500 pulses/s was found to be an optimal smoothness. The filter shapes of the second-order Butterworth lowpass filters and the fourth-order octave filters were similar, however, the octave-filtered velvet noise had much lower ratings.

A repeated measures two-way analysis of variance (ANOVA) test with factors of center or cutoff frequency and repetition was used to assess the statistical significance of the octave-filtered velvet noise and the lowpass-filtered velvet noise separately. the second factor of repetition was tested to determine if there were significant differences in how a participant rated the same stimuli. For both filtered velvet noises, the dependent variable was pulse density. ANOVA preconditions such that the data is normally distributed, the dependent variable of rated pulse density is on a continuous scale and the sphericity of the two trials per participant showing equal amounts of variance were met. More specifically, a Kolmogorov–Smirnov test was conducted on the rated pulse densities for each center and cutoff frequency at a 1% significance level which found the data to be normally distributed for both the lowpass filtered data and the octave filtered data separately. A Mauchly sphericity test was also conducted which found that for each participant, the density ratings across the center or cutoff frequencies were normally distributed.

For the octave-filtered noise, the repeated measures two-way analysis was chosen to show the effect of center-frequencies and repeating trials on participant pulse density rating. The two-way ANOVA test showed that center-frequencies was significant and repeated trials were not significant on pulse density with F-statistics of $F(9, 99) = 79.801, p < .001$ and $F(1, 11) = .995, p > .001$, respectively. The interaction between frequency and repetition was also not significant with an F0-statistic of $F(9, 99) = 1.609, p > .001$.

For the lowpass-filtered noise, the two-way analysis was chosen to show the effect of cutoff frequency and repetition of conditions on pulse density ratings. There was no significant effect from repetition, $F(1, 16) = .278, p > .001$. or from the interaction between repetition and frequency, $F(9, 144) = 1.475, p > .001$. The two-way repeated measures ANOVA testing found only significant effect from cutoff frequencies on pulse density rating such that F(9, 144) = 354.15, $p < .001$.

Additionally, a post-hoc paired t-test was conducted with a Bonferroni-Holm correction on both the lowpass and octave filtered noise tests which showed that between neighboring center or cutoff frequencies, the pulse density ratings were statistically significant ($p < .01$) except between 4 kHz and 8 kHz as well as between 8 kHz and 16 kHz. This means that pulse densities above 4 kHz show no important differences in pulse density ratings.

6. CONCLUSIONS

We studied the frequency-dependency of temporal roughness of velvet noise in human noise perception. The results showed that second-order lowpass filtering and fourth-order octave filtering, the shape of the filter, allows for temporal smearing meaning that the resulting signal can still seem smooth at lower densities than that of broadband velvet noise. Octave-filtered noise at the lowest frequencies showed even lower pulse density ratings than lowpass filtered velvet noise. This study showed that artificial reverberation modeling which utilizes filtering and velvet noise can employ lower pulse densities than what is currently used which allows for more efficient computation.

7. ACKNOWLEDGMENTS

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