RIRS PROCESSING SOFTWARE DEVELOPMENT

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The present work introduces software suitable for acquiring acoustical parameters of a room. This software can provide the user with LSS as required and the ability to process measurements in bulk. The results are stored in an .xlsx file, and the RI analyzed are plotted and saved in .jpg format by band. The option to use octave and third-octave band filters is available. Results shows that over the 250 Hz the difference for EDT, RT20 and RT30 are 16% ± 10% compared to Aurora. Meanwhile, the 125 Hz band has a difference of 17% ± 24%. Below this frequency, the error is significantly higher. Similarly, C50, C80, D50, and cT exhibit noticeable differences below the 125 Hz band. Due to a lack of inter-aural measurements, it is not possible to provide results regarding the IACC.

Keywords: impulse response, acoustical parameters, RI software.

1. Introduction

In order to characterise a room based on its acoustic properties, ISO 3382 develops measurement methods to standardise the process. These methods include the MLS (Maximum Length Sequence) and interrupted noise, as well as the use of an impulsive source such as firecrackers or guns. These methods became obsolete with the development of the LSS (Logarithmic Sine Sweep). The software developed takes LSS measurements as input and derives the acoustic parameters from them.

The obtained parameters include C50, C80, EDT, RT20, RT30, SNR, D50, D80, cT, EDTt, and IACCEarly for both the unfiltered and filtered signals. The option to use octave and third-octave band filters is available to the user. Additionally, for each band, the software provides a graph of the obtained and analyzed IR (impulse response).

The Aurora plugins (available for Adobe Audition and Audacity) are among the most widely used tools for conducting such analysis. Therefore, the development of this work will compare results with that software. Furthermore, the main purpose and usage of this development is the ability to process large quantities of impulse responses automatically, with minimal user intervention. Therefore, the results are saved in well-known formats as are Excel '.xlsx' and '.jpg' files.

2. State of art

Often referred to as "the father of architectural acoustics," Wallace Sabine defined the first acoustical parameter, the RT (Reverberation Time), in 1922 [1]. It is considered the most important characteristic regarding the acoustics of a room. The RT is the time it takes for a sound to decay by 60 dB, which is why it is often called RT60. This is because the dynamic range of a symphony is typically around 60 dB, so it is related to the time it takes for a sound to become inaudible. As it is difficult to have a sound source that generates a 60 dB level above the noise floor, it is commonly used to measure the double of the time it takes for a source to decay by 30 dB, which is referred to as RT30.

Similarly, if it is not feasible to achieve this level of signal-to-noise ratio (SNR), the triple of the time it takes for the signal to decay by 20 dB is measured, known as T20.

Despite the mention in Sabine's notes of the idea that an excessively long reverberation time affects clarity, it was not until 1953 that Thiele defined the parameters D50 and D80, also known as definition for speech and for music, respectively. With this parameters later C50 and C80 were defined.

In 1970, as a complement to the concept of reverberation time, Jordan defined EDT (Early Decay Time) [2].

In the following years, with the advent of computers and their increasing processing power, research linking objective parameters with subjective evaluations became possible. In this field, Yoichi Ando defined that the preference for a room can be characterized by four parameters: listening level, early reflection delay, reverberation time, and interaural cross-correlation [3] [4].

Acoustic parameters continued to be developed over the years, with some of them taking different approaches from the ones mentioned. While initially the focus was on defining a room and predicting the sound field within it for audience preference, in the 1980s, parameters targeting speech intelligibility such as RASTI emerged. Today, efforts are made to evaluate acoustic qualities in diverse environments like stages [5] known as stage parameters, asserting that those parameters are belong to certain boundaries and contexts. All these parameters can be obtained by analyzing the characteristics of the system composed of the sound source and the room itself. As a result, a large number of parameters can be derived solely by characterizing the room through its impulse response. That's why the development of measurement techniques to obtain an accurate RIR (room impulse response, or simply IR) was conducted in parallel with the aforementioned research investigations. To obtain the room IR accurately and, therefore, its associated parameters, it is necessary to develop measurement techniques and mathematical expressions that minimize errors. Among the early methods that contributed to this approach, its inevitable to mention the technique developed by Schroeder [6], which allows for the determination of the decay curve of a signal over time.

In 2009, ISO 3382 was published, standardizing the methods for measuring the reverberation time of a room. This standard describes two methods: MLS and interrupted noise. These methods avoid the use of impulse sources to prevent overloading the measuring instruments. In this standard, acoustic parameters are also described, along with instructions on how to obtain them from the impulse response of the system, which will be detailed in the next section.

In 2007, Angelo Farina described a method for obtaining room impulse responses using a sine sweep and highlighted its advantages compared to the previously mentioned methods. Although this method is not described in ISO 3382, it is widely used for most measurements.

The software described in this work follows Farina's instructions to obtain IR with a LSS.

3. Theoretical Framework

This section briefly introduces the justification of using the LSS as a novel method for obtaining the IR of a system. Once obtained, acoustical parameters are calculated according to ISO 3382 specifications.

3.1 Systems

A system is a process that, upon receiving a signal at its input, produces another signal at its output. This system can be physical, such as a mass-spring system, or an abstract entity, such as a mathematical expression.

In the field of architectural acoustics, the system under evaluation is often the physical room itself. Figure 1 represents the system as $h(t)$, which describes the relationship between the input signal $x(t)$ and the output signal $y(t)$ in the time domain.

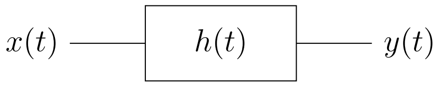

Figure 1: System representation.

To obtain $y(t)$, it is necessary to convolve $x(t)$ and $h(t)$ in the time domain. Alternatively, in the frequency domain, it is possible to apply the product of $X(\omega)$ and $H(\omega)$, as shown in eq.1:

$$y_{(t)} = x_{(t)} * h_{(t)} = \int_{-\infty}^{+\infty} x_{(t)} h_{(t-\tau)} dt \tag{1}$$

While in frequency domain, eq.and 2 shows $Y(\omega)$:

$$Y(\omega) = X(\omega) * H(\omega) \tag{2}$$

Thus, the relationship between input and output is the system itself in eq. 3.

$$H(\omega) = \frac{Y(\omega)}{X(\omega)} \tag{3}$$

If the input is 1, the relationship between $H(\omega)$ and $Y(\omega)$ is simplified, this particular case is called 'impulse response'.

3.2 Impulse Response

The ideal impulse is called the Dirac delta function. It is defined such that its area under the curve is equal to 1, meaning its integral over time is 1. In the limit as the duration in time approaches zero ($t \to 0$), the amplitude of the Dirac delta function must tend towards infinity ($A \to \infty$).

It is important to note that the Dirac delta function itself does not exist physically and cannot be directly realized in practice. eq.4 describes it.

$$\delta(x) = \begin{cases} \infty, & \text{si } x = 0 \\ 0, & \text{si } x \neq 0 \end{cases} \tag{4}$$

Thus, its integral is as described in Eq. 5:

$$\int_{-\infty}^{\infty} \delta(x)\, dx = 1 \tag{5}$$

By definition, its Fourier transform is 1 for all the spectrum, shown in eq. 6:

$$\mathcal{F}[\delta(t)] = 1 \tag{6}$$

Therefore, the signal in the time and frequency domains is illustrated in figure 2:

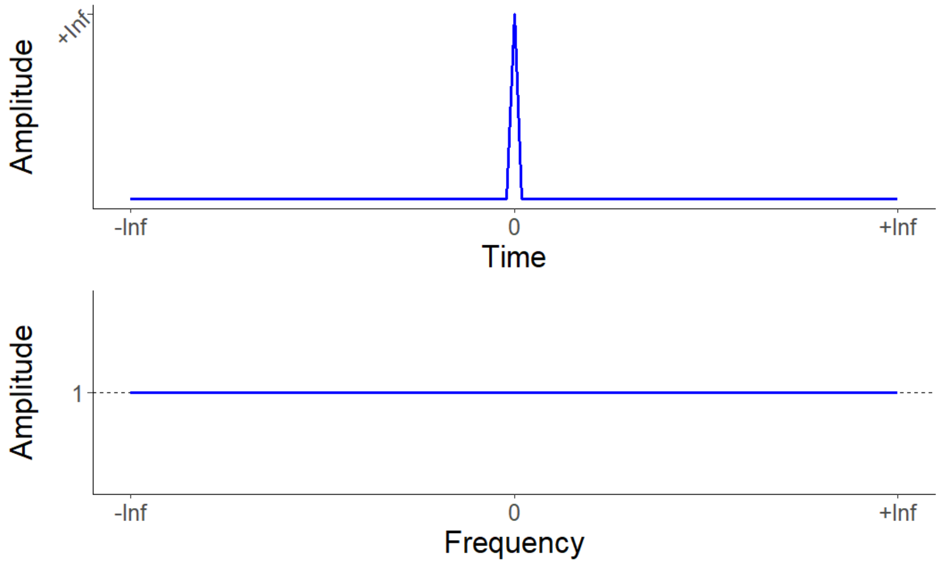

Figure 2: Delta Dirac represented in time and frequency domain.

3.3 Logarithmic Sine Sweep

Since obtaining an infinite spectrum is not possible, it needs to be limited, in amplitude and spectrum.

A bounded spectrum will generate an impulse that will not be defined as the Dirac delta. It will have a longer and realistic duration in time. Additionally, its spectrum will no longer be constant but rather a sinc function, shown in figure 3

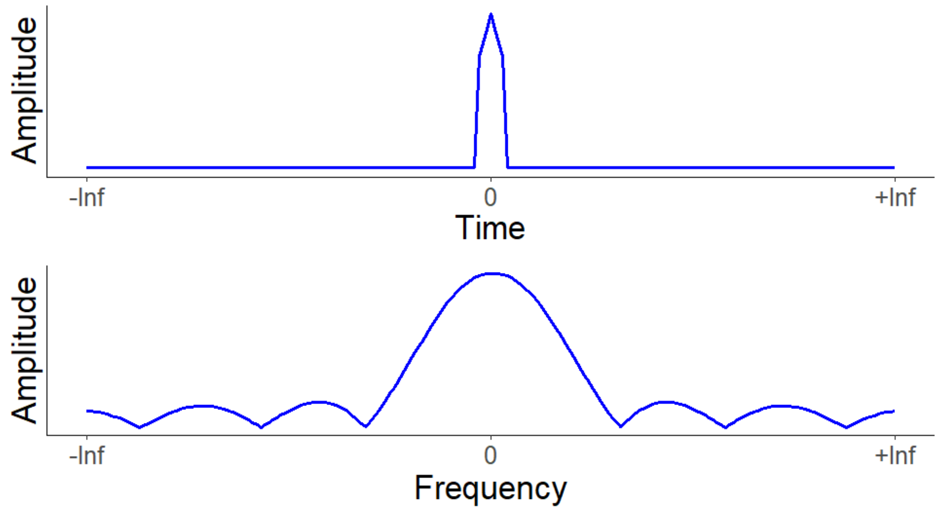

Figure 3: Sinc. Limited in spectrum and amplitude.

Since the Dirac delta function has a value of 1 in the frequency domain, it can be understood that any signal with a multiplicative inverse in the frequency domain will result in a Dirac delta function in the time domain. Based on this premise, LSS was developed. It consists of a sine wave that gradually increases in frequency, spanning the entire required spectrum. The same amount of time is allocated to traverse each frequency band. The signal is then processed with its inverse to obtain the desired Dirac delta impulse stimulus. In 2007 Farina [7] presented the LSS defined by eq. 7:

$$LSS_(t) = sin\left(\frac{\omega_1.T}{ln(\frac{\omega_2}{\omega_1})}\left(e^{\frac{t}{T}ln(\frac{\omega_2}{\omega_1})} - 1\right)\right) \tag{7}$$

ω_1 is the starting frequency, ω_2 the end frequency. Duration is T.
The signal generated and its spectrum is shown in fig. 4:

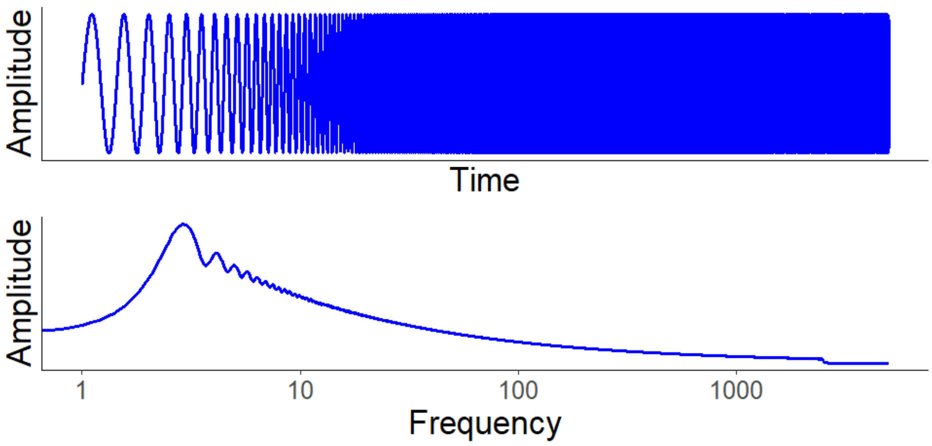

Figure 4: LSS in time and frequency domain.

As can be seen, the energy by band is not constant. Then, its inverse of the LSS must be as fig. 5 shows:

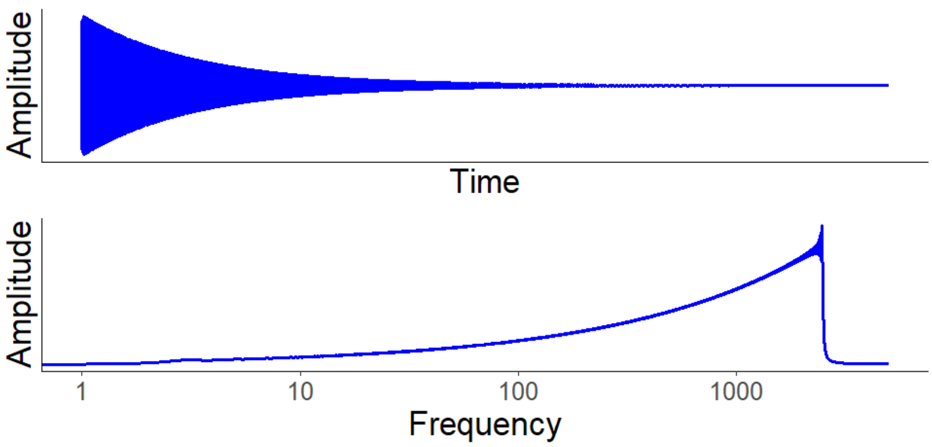

Figure 5: LSS inverse represented in time and frequency domain.

Then, the spectrum of the convolution of these two signals is as figure 6:

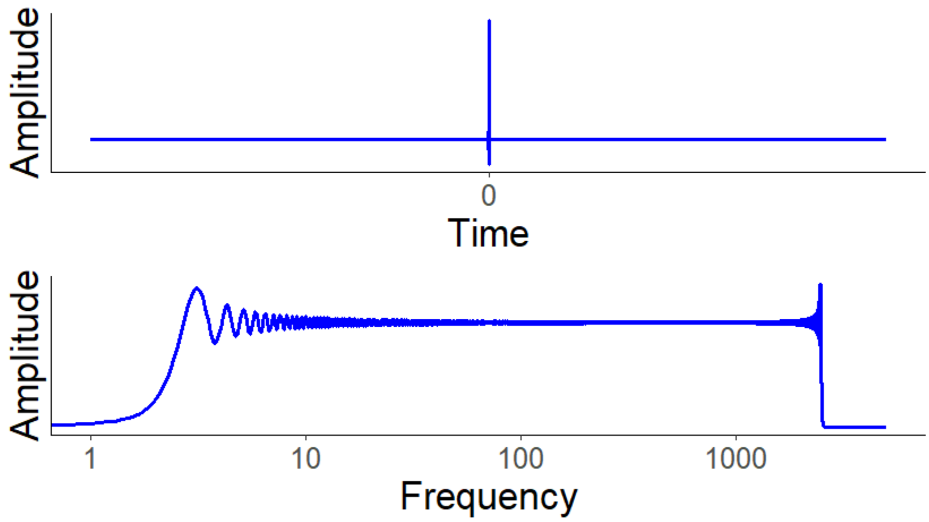

Figure 6: Convolution of the LSS and its inverse represented in time and frequency domain.

To avoid peaks and ringing in the spectrum a fade-in fade-out is applied to the signal, the results of using these two signals are shown in fig. 7:

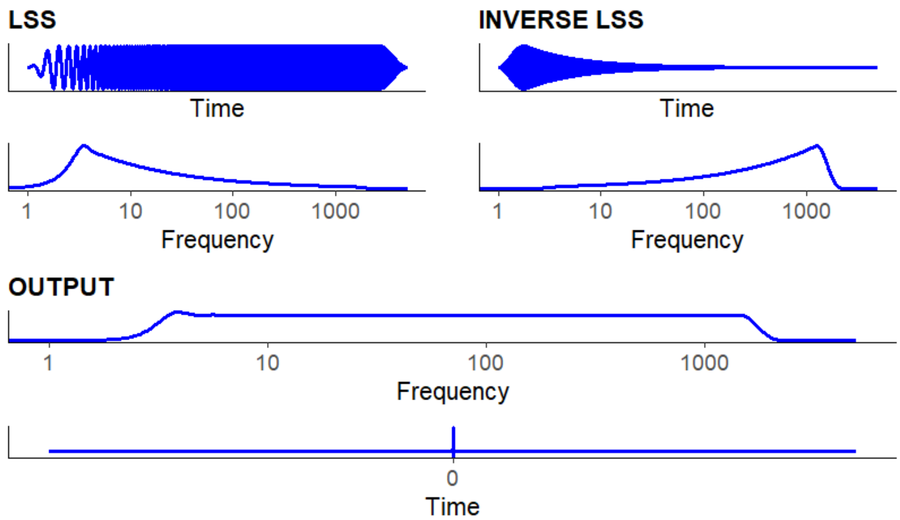

Figure 7: Effect of the spectrum when a fade-in fade-out is added to the LSS.

3.4 Acoustical Parameters

Using LSS and its inverse to measure a limited-band impulse response of the room, further analysis can be made.

Reverberation Time

Often called RT60, is the time required for a sound to decay by 60 dB after the sound source has ceased. By definition, the source has to be able to generate an output 60 dB above the noise floor, and since this is not possible most of the times, RT30 and RT20 are defined.RT30 is the time that it takes a signal to decay 30 dB, doubled. While RT20 is the time the signal decays 20 dB, tripled.

Analyzing the decay curve of the IR, RT60 is the difference in time between -5 dB and -60 dB. While RT30 is the difference between -5 dB and -30 dB, and RT20 is the difference between -5 dB and -20 dB. RT60 is considered the most important descriptor for room preference.

Early Decay Time

EDT is the time difference between 0 dB and -10 dB, multiplied by 6. Compared to RT60, provides a measure of the early decay of the curve. Related to the early reflections, and thus, the sensation of clarity, spaciousness and intimacy. Figure 8 shows the relationship between EDT, RT60, RT30 and RT20. L1 is the level of the sound source, and L2 the noise floor.

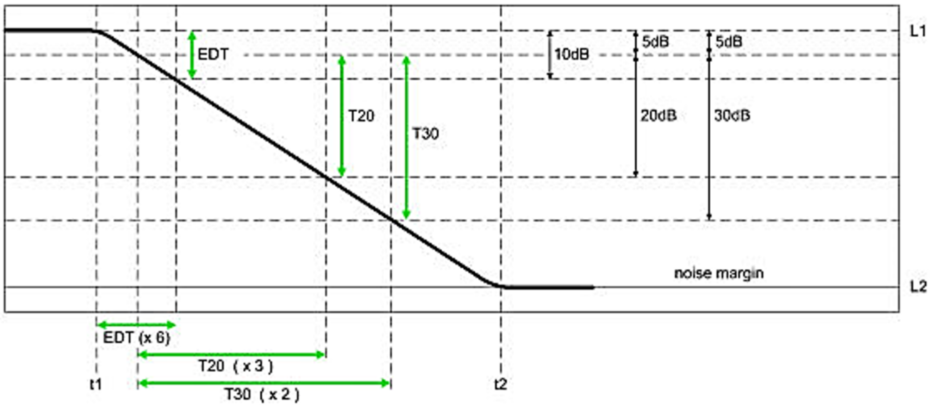

Figure 8: LSS in time and frequency domain.

Clarity and Definition

These parameters are defined as eq. 8 and eq. 9 shows:

$$C_{t_e} = 10 \log_{10} \frac{\int_0^{t_e} h_{(t)}^2 dt}{\int_{t_e}^{\infty} h_{(t)}^2 dt} dB \tag{8}$$

Being t_e the early-to-late index. For music is 80 ms and for speech 50 ms.

$$D_{t_e} = \frac{\int_0^{t_e} h_{(t)}^2 dt}{\int_0^{\infty} h_{(t)}^2 dt} dB \tag{9}$$

Both are related via eq. 10

$$C_{t_e} = 10 \log_{10} \left(\frac{D_{t_e}}{1 - D_{t_e}} \right) \tag{10}$$

Centre time and EDTt

Center time is the center of mass of $h(t)$, and its obtained as eq. 11 shows:

$$T_s = \frac{\int_0^{\infty} t\, h_{(t)}^2 dt}{\int_0^{\infty} h_{(t)}^2 dt} dB \tag{11}$$

It provides a division between early and late reflections. With this parameter, EDT_t is defined as the time the decay reaches -10 dB after the centre time, multiplied by 6.

Inter-aural Cross Correlation IACC

It cuantifies the similarity between sound arriving at left and right ears. Its defined as shown in eq. 12:

$$IACC = max \left| \frac{\int_{t_1}^{t_2} h^2_{left(t)} \; h^2_{right(t+\tau)} dt}{\sqrt{\int_{t_1}^{t_2} h^2_{left(t)} dt \; \int_{t_1}^{t_2} h^2_{rigth(t)} dt}} \right| \tag{12}$$

With $|\tau| < 1ms$. IACC Early assumes $t_1 = 0s$ and $t_2 = 80ms$.

4. Software Development

The language used to develop this software is R. Mainly because is easy to install, it only requires R and RStudio to run in most of computers with Windows, MacOS and Linux.

Also, the UI is simplified by using Shiny packages that allows to run script directly on the default web browser of the computer.

4.1 UI

The interface consists of 3 tabs. Each tab requires inputs from the user, and there is a brief explanation on how to use it. The first tab allows the user to create their preferred LSS. They have the option to choose the minimum frequency, maximum frequency, duration, and sampling frequency. Additionally, the user must enter the working directory where the LSS.wav and inverse filter will be saved. First and default tab is shown in figure 9

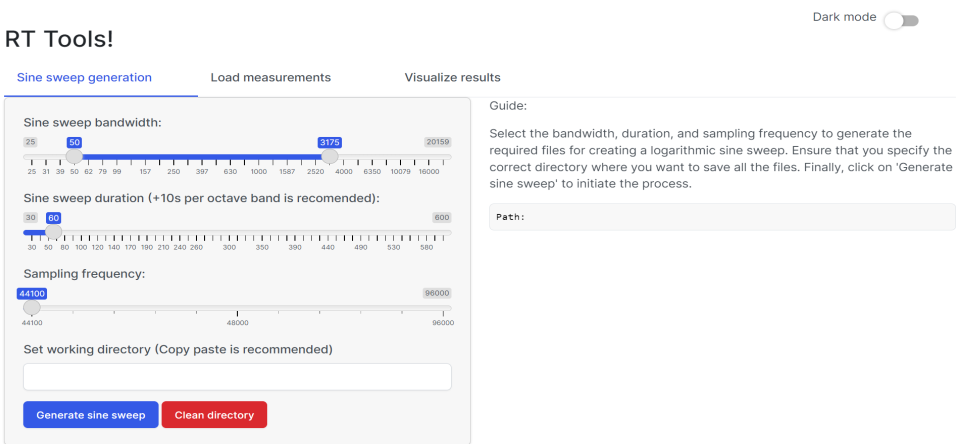

Figure 9: First of tree tabs in the UI.

Once the user clicks ’Generate sine sweep’, in the folder chosen tree files are created. Sweep.wav, contained the measurement signal. fInv.wav, which will be later used to obtain the IR. And data generated.xlsx containing frequency range and sampling frequency of the file created.

On the second tab, the user must provide a directory path containing the ’fInv.wav’ and all ’.wav’ files of the measurements to analyze. Second tab is shown in figure 10

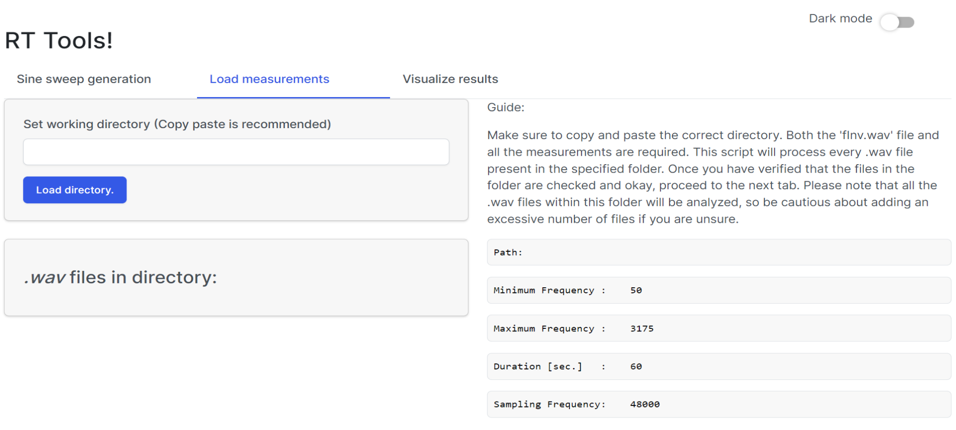

Figure 10: Second of tree tabs in the UI.

Once the program checks the presence of the inverse filter and other measurements in the folder, it prompts the user a notification to proceed to analyze all the data within that folder. 'filelist.xlsx' is created, which includes all file names found for analyze.

In the third and final tab the user can choose to generate plots. If the IR is too short (less than 2 seconds) there is the option to choose shorter time vectors for the plots in order to improve the visualization of the results. The option for using third octave band filters is available.

4.2 Data generation

In the first tab, the user is provided with the LSS signal and its inverse filter. These signals are generated using the following lines of code:

```
# Time vector.
time  <- seq(0, duration, length.out = Fs * duration)
# Auxiliary value
R   <- log(f2 / f1)
# Compensation for the inverse filter.
k   <- exp(time * R / duration)
# LSS signal. With no fade-in fade-out smooth,
LSS <- sin(2 * pi * f1 * duration / (R) * (k - 1))
# Inverse LSS signal with fade-in fade-out smooth.
I_LSS   <- sweep[length(sweep):1] / k * fade
# Smooth of the LSS and normalization.
LSS <- sweep / (max(abs(sweep))) * (2 ^ 15 - 1) * fade
# Normalization of the I_LSS
I_LSS   <- fInv / max(fInv) * (2 ^ 15 - 1)
```

Once the measurements are done, the files must be placed in the same folder as the inverse filter. Then, the code to generate the impulse response is as follows:

```
# Read I_LSS and its sampling frequency.
I_LSS   <- readWave("I_LSS.wav")@left
Fs1     <- readWave("I_LSS.wav")@left
# Read the filename (measurement)
signal <- readWave(filename)@left
Fs2     <- readWave(filename)@left
# If the sampling frequencies do not match, the measurement is resampled.
if (Fs1 != Fs1) { signal <- signal::resample(signal, Fs1, Fs2) }
```

```
# If signal and I_LSS lengths do not match add zero padding.
max_length <- max(length(I_LSS), length(signal))
I_LSS <- c(I_LSS, rep(0, max_length - length(I_LSS)))
signal <- c(signal, rep(0, max_length - length(signal)))
# Due to the multiplication of the signals in the spectral domain, the
# output will always begin with the start of the impulse response at
# the first sample. Assuming that RI is always less than 6 seconds.
# the remaining part of the signal is discarded for efficiency.
RI <- convolve(I_LSS, rev(signal), conj = TRUE)[1:(6 * Fs)]
# Note that R function convolve does provide the convolution of the
# signal, but it does not calculate the result in the time domain,
# but in the frequency domain.
```

Before proceeding with calculations, is necessary to get the IR by bands:

```
for ( i in 1:ncol(RI)){
  # In each column of the RI dataframe are the IR of every file in folder.
  signal <- RI[,i]
  Fs <- readWave(fileNames[i,1])@samp.rate
  # bandsFilter is the list with all the frecuencies by band defined in
  # UNE-EN 61260. The bandsFilter[1,3] depends on the sampling frequency.
  bandsFilter[1,3] <- Fs/2
  RI_aux <- NULL
  for ( j in 1:nrow(bandsFilter) ) {
    # The signal is reversed to minimize the effects of the filter, but
    # it  does not improves results since seewave::filter uses FIR filters.
    aux <- seewave::ffilter( rev(signal), Fs,
                              from = as.numeric(bandsFilter[j, 1]),
                              to = as.numeric(bandsFilter[j, 3]),
                              rescale=FALSE
               )
    RI_aux <- cbind(RI_aux, rev(aux))
  }
  RIbyBand[[i]] <- RI_aux
}
```

With the IR for each frequency band and global is now possible to calculate the parameters described in the theoretical framework. To evaluate the IR, it is necessary to smooth the signal and apply a logarithmic scale to it.

```
# Smooth the signal using a moving average with a window  size that depends
# on the maximum frequency of the signal, Fmax. A window size of (Fs/2*FMax)
# for a moving average filter works as a low-pass filter with a cutoff
# frequency at Fmax.
signal <- as.numeric(
          rollmean( rev(signal),
                    k = round(Fs/(2*Fmax)),
                    align = "right")
        )
signal <- rev(signal)

# Apply Hilbert transform and obtain the envelope of the smooth signal,
# keep the real part only.
smooth01 <- Re(seewave::env((signal), Fs, plot = FALSE))
# Convert the envelope into decibels and normalize it to have a maximum
# of 0dB.
smooth02 <- 10*log10((smooth01^2)/ sum(smooth01^2))
smooth02 <- smooth02 - max(smooth02)
```

```
# Noise floor is asumed to be in the last 25% of the RI.
noise_floor <-  mean(smooth02[(length(smooth02)*0.75):
                             (length(smooth02))] )
# Find the first local minima over the smooth signal. This assumes a
# certain degree of diffusive field and high enough SNR in the measurement.
minima <- max(smooth02)
# The objective of this block of code is to obtain the sample at which the
# signal reaches the noise floor. The signal is divided into several lengths,
# and if the mean of the current block of signal is higher than the mean of
# the next block of signal, a new minimum value is found, indicating that
# the signal is decaying. If not, then the last value found is considered
# the actual minimum value, indicating that the noise floor has been reached.
# To avoid outliers, these mean values are calculated multiple times using
# different lengths (seq(0.01, 0.1, 0.01)) and compared to the minimum value
# of the next block of signal. The new_min is the sample where the signal
# reaches noise floor. 'minima' is the minimum value found.
for(j in seq(0.01, 0.1, 0.01))    {
    step_min <- Fs*j
    flag <- 0
    for (i in seq(which.max(smooth02),length(smooth02)-step_min, step_min)){
        if (minima > mean(smooth02[i:(i+step_min)]) & flag == 0){
            new_min <- i
            minima  <- mean(smooth02[i:(i+step_min)])
        }
        if (minima < mean(smooth02[i:(i+step_min)])){
            if (minima < (noise_floor)) flag <- 1
        }
    }
}
# Compute the reverberation time parameters EDT, RT20, and RT30 using
# the smooth signal with the Schroeder's integral.
smooth03 <- smooth01[1:(new_min)]
smooth03 <- rev(10*log10(cumsum(rev(smooth03^2)/sum(smooth03^2))))
EDT  <- (which(smooth03 <= -10)[1] - which.max(smooth02))       * 6 / Fs
RT20 <- (which(smooth03 <= -25)[1] - which(smooth03 <= -05)[1])* 3 / Fs
RT30 <- (which(smooth03 <= -35)[1] - which(smooth03 <= -05)[1])* 2 / Fs
# Compute the clarity parameters C50 and C80, and definition D50
# using the signal within the RT window with aux vector.
aux <-  signal[(which.max(smooth02)):new_min]^2
C50  <- 10*log10(sum(aux[1:(0.050*Fs)])/ sum(aux[(0.050*Fs):length(aux)]))
C80  <- 10*log10(sum(aux[1:(0.080*Fs)])/ sum(aux[(0.080*Fs):length(aux)]))
D50  <- sum(aux[1:(0.050*Fs)])/ sum(aux[1:length(aux)])*100
# Centre time sample:
cT   <- round(sum(1:length(aux) * aux) / sum(aux))
EDTtsignal   <- smooth03[(cT):length(smooth03)]
EDTt  <- (which(EDTtsignal <= -10)[1] - which.max(EDTtsignal))* 6 / Fs
SNR  <- -noise_floor
# Centre time from samples to time:
cT <- cT/Fs*1000
```

The parameters IACC is only done when there are only two files in folder and its calculations are done in a different block.

```
# RIbyBand[[1]] is the left and RIbyBand[[2]] right.
start <- which.max(RIbyBand[[1]][, 1])
if (start < Fs/1000) start <- Fs/1000        # Sanity check to avoid errors.
end <- start + 0.080 * Fs                    # IACC early 0 to 80 ms.
paramIACC <- numeric(dim(RIbyBand[[1]])[2]) # Initialize results dimension.
for(i in 1:dim(RIbyBand[[1]])[2]){
```

```
    # Left is loaded with the time window defined by the global signal.
    left  <- RIbyBand[[1]][,i][start:end]
    # Divisor of the IACC function does not change with tau.
    div <- sqrt(sum(left^2) * sum(RIbyBand[[2]][,i][start:end]) )
    aux1 <- 0
    aux2 <- 0
    # tau from -1 ms to 1ms (ISO 3382)
    for ( tau in -round(Fs/1000):round(Fs/1000)){
      right <- RIbyBand[[2]][,i][(start:end)+tau]
      aux1 <- sum( left * right ) / div
      if(aux1 > aux2){ aux2 <- aux1 }          # Keep the max value found.
    }
    paramIACC[i] <- aux2
  } return(paramIACC)
```

4.3 Output

Since the script is designed to process a batch of measurements, multiple plots (11 for octave band filters and 31 for third-octave band filters) as well as tables (1 for each file, plus average and standard deviation) are created. Although this data is not displayed in the GUI, it is saved in the working directory. Figure 11 shows an example of an unfiltered signal:

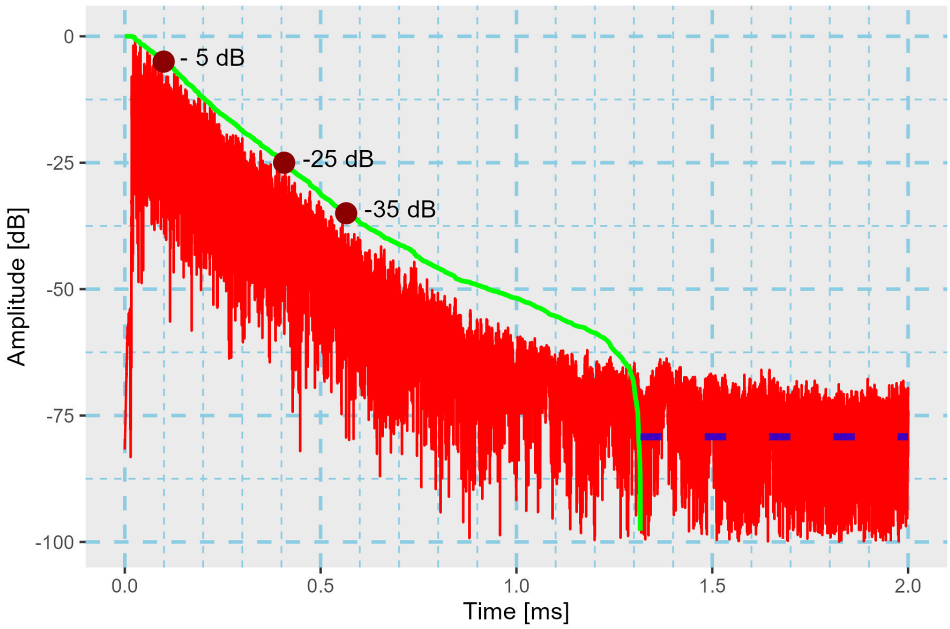

Figure 11: Global.jpg, output example.

The points used to calculate RT are marked with dark red dots. The smooth measurement signal is displayed in red, and the obtained decay curve is shown in green.

When evaluating only two signals, the output of the .xlsx file has 4 sheets. Standard deviation, mean, and the individual results for those measurements.

	Global	31.5	63	125	250	500	1000	2000	4000	8000	16000
C50	-4.973	0.915	0.969	1.902	-2.716	-0.736	-5.128	-3.649	-1.054	1.518	-5.723
C80	-2.474	2.653	2.566	5.955	0.206	3.141	-2.777	-1.640	1.687	4.868	-1.638
EDT	3.325	0.877	0.898	0.794	1.569	1.139	1.470	2.471	1.050	0.846	-0.407
RT20	1.727	1.963	1.960	1.630	1.657	1.601	1.016	1.423	0.925	4.898	1.490
RT30	1.475	2.691	2.575	1.652	1.731	1.442	0.947	1.206	0.828	3.933	1.092
SNR	67.520	68.946	69.696	79.844	89.094	72.657	62.024	66.322	68.060	94.574	50.557
D50	29.660	55.181	55.512	60.801	34.904	45.787	32.134	33.842	44.404	58.653	31.827
D80	40.895	64.593	64.181	79.691	51.192	67.001	41.581	43.235	58.247	75.411	46.219
cT	959.740	80.729	81.469	63.062	123.885	86.448	491.156	587.198	176.417	94.927	337.990
EDTt	0.385	0.596	0.620	0.588	0.965	0.730	0.303	0.385	0.322	0.385	12.113
IACC	0.249	0.922	0.922	0.804	0.611	0.263	0.157	0.167	0.123	0.144	0.012

Average | Standard deviation | Earthworks 1 - med 1.wav | Earthworks 2 - med 1.wav

Figure 12: Output.xlsx, output example for two files.

Can be seen that sheets 'Average', 'Standard Deviation' and the ones corresponding to the measurements are created. Also, when two files are analyzed IACC is attached.

5. Results and Discussion

The signal used in the analysis is limited to the frequency range of 35 Hz to 15 kHz. Therefore, any results obtained beyond this frequency range are not taken into account or considered for the analysis.

The results are promising when comparing with Aurora. There is minimal difference between the two programs above 125 Hz. As mentioned before, this script is designed to analyze multiple measurements simultaneously. In this case, the average results will be compared with each other for five sets of 12 measurements, specifically in octave bands.

Table 1 shows the difference of the mean values obtained with Aurora and the R script over 5 sets of 12 measurements. Bold values corresponds to the ones above the JND defined in ISO 3382 for C50 (JND for C80 is assumed to be the same of C50 just to highlight differences), D50 and EDT. For RT the JND used is 20 %, defined by Blevins [8].

Table 1: Mean difference between Aurora and R script.

Freq.[Hz]	Lin.	31.5	63	125	250	500	1 k	2 k	4 k	8 k	16 k
C50	0.68	0.58	0.94	0.46	**1.07**	0.53	0.61	0.37	0.29	0.12	0.41
C80	0.40	0.08	0.23	0.30	0.28	0.19	0.37	0.27	0.26	0.12	0.77
EDT[%]	5	**26**	5	17	14	16	12	9	7	9	19
RT20[%]	3	5	**28**	15	3	1	1	1	0	1	13
RT30[%]	3	**34**	**44**	13	2	1	1	1	0	1	18
D50[%]	3.90	2.99	**5.22**	2.62	**5.58**	3.21	3.53	1.72	1.58	0.44	1.32
cT[ms]	4.1	**50.0**	**21.5**	7.7	**12.2**	2.7	1.9	1.5	1.5	0.3	1.4

It can be observed that the differences are acceptable in all frequency bands except for the low frequencies (31.5 Hz and 63 Hz). At these frequencies, the differences for RT20, RT30, and EDT are significantly larger. This behavior is expected since the analysis of data with Aurora at these specific bands results in RT20 and RT30 differences ranging from 0.65 to 1.2 seconds, between these parameters for the same file and group. Whereas the differences at higher frequencies are smaller than 0.06 seconds. The impact of these differences can be noticed in the decay curve. Figure 13 illustrates the decay at 31.5 Hz:

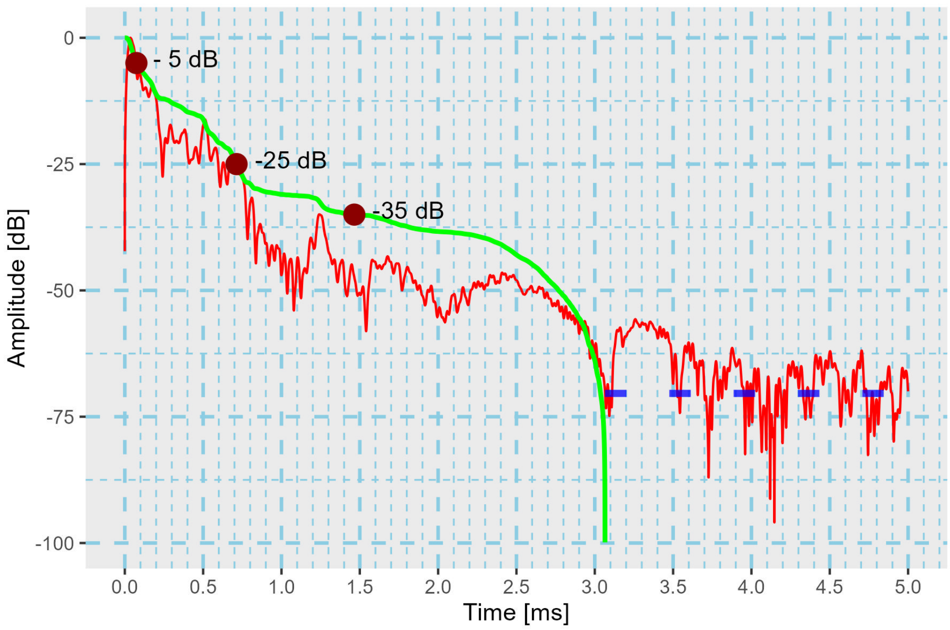

Figure 13: Decay obtained at 31.5 Hz with R script.

At first glance, the differences in RT20, RT30, and EDT are noticeable. It is worth noting that this decay pattern is consistent across the majority of the evaluated measurements. Next table, table 2, presents the standard deviation of the comparison results. Low values indicate a high level of consistency among the results. These table is essential for validating the developed script.

Table 2: Standard deviation

Freq.[Hz]	Lin.	31.5	63	125	250	500	1 k	2 k	4 k	8 k	16 k
C50[dB]	0.1	0.0	0.2	0.2	0.2	0.4	0.2	0.2	0.1	0.1	0.2
C80[dB]	0.2	0.0	0.1	0.2	0.3	0.2	0.1	0.1	0.1	0.1	0.1
EDT[%]	1.8	0.2	1.6	1.5	2.6	5.1	2.3	1.3	0.7	0.5	1.3
RT20[%]	2.9	3.6	6.0	**9.7**	2.2	0.4	0.3	0.2	0.4	0.5	1.8
RT30[%]	4.2	6.8	4.9	**11.9**	0.9	1.0	0.4	0.3	0.3	0.5	2.5
D50[%]	0.6	0.3	0.9	1.0	1.4	2.3	1.0	0.9	0.5	0.3	0.5
cT[ms]	1.6	0.5	1.9	2.5	1.5	2.0	1.1	0.9	0.7	0.3	0.2

For the EDT, RT20, and RT30 it can be observed from the standard deviation that above 250 Hz, the variability of the results is minimal. The 125 Hz band shows high variability for RT20 and RT30, indicating the need to further develop and evaluate the signal in that frequency range. The results suggest that similar considerations should be taken into account for the 31.5 Hz and 63 Hz bands. The remaining parameters do not exhibit variability that significantly affects the results when compared to their associated JND values. Consequently, it is not advisable to utilize the developed script for individual file analysis, but rather for bulk analysis purposes.

6. Conclusions

The work carried out meets the stated objectives. A script was developed in R that processes a large number of measurements without the need for user intervention. This processing is saved in an .xlsx file, and if the user requires it, the corresponding octave or third-octave band graphs will be generated.

As it was designed to process a large number of measurements, the comparison of 5 groups of 12 measurements showed acceptable differences with the Aurora software. For frequencies between the 250 Hz and 8 kHz bands, the difference was found to be $14\% \pm 10\%$, which corresponds to the proposed JND values mentioned earlier, for the parameters of EDT, TR20, and TR30.

The 125 Hz band needs to be handled with caution as the associated error can be as high as $17\% \pm 24\%$. For the 63 Hz and 31.5 Hz bands, the error is well above the suggested JND, as high as $44\% \pm 12\%$, indicating the need to continue developing and working on filters for those frequencies. The use of filters from other packages can be considered. Additionally, contacting Angelo Farina to inquire about the method used to smooth the signal in those bands is an option. This may lead to better results, not only to minimize differences with Aurora but also to evaluate if the process used aligns with the approach proposed in Farina's work. Another achieved objective is the development of software that runs without the need for the user to make too many configurations on their computer. Shiny provides a server that runs on the local host and can be accessed from any browser, without the need of installing software. Although it still needs to be tested on other systems, installing the R interpreter and running an auxiliary script to install the necessary packages should be sufficient to run the software.

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7. Appendix

8. Results

Five sets of data were analyzed.

Set 1

AURORA	Global	31.5	63.0	125.0	250.0	500.0	1.0k	2.0k	4.0k	8.0k	16.0k
"C50"	1.2	−0.7	−1.1	0.9	−1.0	0.8	1.8	0.9	1.0	2.7	7.2
"C80"	4.5	2.3	2.3	3.6	2.8	4.1	5.0	4.1	4.3	6.3	12.1
"EDT"	0.9	1.4	1.1	1.0	0.9	0.9	0.8	0.9	0.9	0.7	0.4
"RT20"	0.9	1.6	1.2	0.9	1.0	0.9	0.9	0.9	0.9	0.7	0.4
"RT30"	0.9	2.2	1.2	1.0	1.0	0.9	0.9	0.9	0.9	0.7	0.4
"SNR"	52.8	31.6	42.0	50.7	54.5	56.1	53.5	54.0	51.7	50.7	53.9
"cT"	59.7	137.6	108.2	79.9	81.4	64.9	57.0	64.2	61.5	47.7	26.8

R SCRIPT	Global	31.5	63.0	125.0	250.0	500.0	1.0k	2.0k	4.0k	8.0k	16.0k
"C50"	" 0.5"	"−0.3"	"−0.2"	" 0.2"	" 0.3"	"−0.3"	" 1.3"	" 0.8"	" 0.9"	" 2.7"	" 7.8"
"C80"	" 3.8"	" 2.3"	" 2.2"	" 3.0"	" 3.5"	" 3.6"	" 4.7"	" 3.9"	" 4.1"	" 6.3"	"13.0"
"EDT"	" 0.8"	" 1.0"	" 1.0"	" 0.8"	" 0.8"	" 0.8"	" 0.7"	" 0.8"	" 0.8"	" 0.6"	" 0.3"
"RT20"	" 0.9"	" 1.6"	" 1.5"	" 1.1"	" 1.0"	" 0.9"	" 0.8"	" 0.9"	" 0.9"	" 0.7"	" 0.4"
"RT30"	" 1.0"	" 1.8"	" 1.9"	" 1.3"	" 1.0"	" 0.9"	" 0.8"	" 0.9"	" 0.9"	" 0.7"	" 0.4"
"SNR"	"79.6"	"53.4"	"54.3"	"65.5"	"76.1"	"78.8"	"78.4"	"80.1"	"78.1"	"80.1"	"78.0"
"cT"	"64.9"	"88.0"	"87.2"	"74.4"	"68.4"	"66.4"	"57.3"	"63.7"	"62.1"	"47.1"	"25.1"

Difference	Global	31.5	63.0	125.0	250.0	500.0	1.0k	2.0k	4.0k	8.0k	16.0k
"C50"	0.7	0.5	0.9	0.7	1.4	1.2	0.4	0.1	0.2	0.0	0.7
"C80"	0.7	0.0	0.1	0.6	0.7	0.5	0.3	0.2	0.1	0.0	0.9
"EDT"	0.0	0.3	0.0	0.1	0.1	0.1	0.1	0.1	0.1	0.1	0.2
"RT20"	0.1	0.0	0.3	0.2	0.0	0.0	0.0	0.0	0.0	0.0	0.1
"RT30"	0.1	0.2	0.6	0.3	0.0	0.0	0.0	0.0	0.0	0.0	0.2
"SNR"	26.8	21.8	12.2	14.8	21.5	22.7	24.9	26.1	26.4	29.4	24.0
"cT"	5.3	49.6	21.0	5.6	13.1	1.5	0.3	0.5	0.6	0.6	1.8

Set 2

AURORA	Global	31.5	63.0	125.0	250.0	500.0	1.0k	2.0k	4.0k	8.0k	16.0k
"C50"	0.5	−0.7	−1.2	0.8	−1.0	−0.1	1.2	0.1	0.3	2.2	6.7
"C80"	3.5	2.3	2.3	3.4	3.0	3.2	3.9	2.8	3.2	5.5	11.4
"EDT"	1.0	1.4	1.1	1.0	1.0	1.0	1.0	1.1	1.0	0.8	0.4
"RT20"	1.1	1.8	1.2	0.9	1.1	1.0	1.0	1.1	1.1	0.8	0.5
"RT30"	1.1	2.5	1.2	1.0	1.1	1.0	1.0	1.1	1.1	0.8	0.5
"SNR"	50.8	29.2	39.0	39.2	48.6	50.2	50.8	52.6	50.8	49.7	56.6
"cT"	69.1	138.2	107.7	80.5	81.4	74.1	66.1	76.3	71.8	52.6	28.5

R SCRIPT	Global	31.5	63.0	125.0	250.0	500.0	1.0k	2.0k	4.0k	8.0k	16.0k
"C50"	−0.1	−0.2	−0.2	0.2	0.1	0.4	0.5	−0.4	−0.1	2.0	7.0
"C80"	3.2	2.3	2.2	3.3	3.2	3.4	3.4	2.5	2.9	5.4	12.2
"EDT"	1.0	1.0	1.0	0.8	0.8	0.8	0.9	1.0	1.0	0.7	0.3
"RT20"	1.1	1.6	1.6	1.1	1.0	1.0	1.0	1.1	1.1	0.8	0.4
"RT30"	1.1	1.9	1.8	1.2	1.0	1.0	1.0	1.1	1.1	0.8	0.4
"SNR"	76.1	52.3	53.0	59.3	70.8	75.7	76.1	79.7	79.0	81.7	77.4
"cT"	72.1	88.5	87.7	73.3	69.4	68.4	68.7	79.0	73.4	53.2	27.2

Difference	Global	31.5	63.0	125.0	250.0	500.0	1.0k	2.0k	4.0k	8.0k	16.0k
"C50"	0.7	0.5	1.0	0.6	1.1	0.5	0.7	0.4	0.3	0.2	0.3
"C80"	0.3	0.0	0.1	0.1	0.1	0.2	0.5	0.4	0.3	0.1	0.7
"EDT"	0.1	0.3	0.0	0.2	0.2	0.2	0.1	0.1	0.1	0.1	0.2
"RT20"	0.0	0.1	0.3	0.2	0.1	0.0	0.0	0.0	0.0	0.0	0.1
"RT30"	0.0	0.2	0.5	0.2	0.0	0.0	0.0	0.0	0.0	0.0	0.2
"SNR"	25.3	23.1	14.0	20.1	22.2	25.6	25.2	27.1	28.2	32.1	20.8
"cT"	3.0	49.7	20.1	7.3	12.1	5.7	2.6	2.6	1.6	0.6	1.3

Set 3

AURORA	Global	31.5	63.0	125.0	250.0	500.0	1.0k	2.0k	4.0k	8.0k	16.0k
"C50"	0.6	−0.7	−1.1	0.9	−1.1	0.2	1.3	−0.1	0.4	2.2	6.8
"C80"	3.5	2.3	2.4	3.5	2.9	3.6	3.9	2.7	3.4	5.5	11.4
"EDT"	1.0	1.4	1.1	1.0	1.0	1.0	0.9	1.1	1.0	0.8	0.4
"RT20"	1.0	1.6	1.1	0.9	1.0	1.0	1.0	1.1	1.0	0.8	0.5
"RT30"	1.0	2.5	1.1	1.0	1.0	1.0	1.0	1.1	1.0	0.8	0.5
"SNR"	46.6	29.8	32.3	46.7	50.4	50.5	45.7	45.3	48.0	48.6	61.4
"cT"	67.8	136.9	106.4	79.5	81.9	71.3	63.8	75.7	70.2	52.8	28.0

R SCRIPT	Global	31.5	63.0	125.0	250.0	500.0	1.0k	2.0k	4.0k	8.0k	16.0k
"C50"	0.0	−0.1	−0.1	0.3	−0.3	−0.2	0.7	−0.3	0.1	2.1	7.2
"C80"	3.3	2.4	2.3	3.1	2.9	3.5	3.5	2.5	3.1	5.5	12.2
"EDT"	0.9	1.0	1.0	0.8	0.8	0.9	0.8	1.0	0.9	0.7	0.3
"RT20"	1.0	1.5	1.5	1.1	1.0	1.0	0.9	1.1	1.0	0.8	0.4
"RT30"	1.1	1.8	1.7	1.1	1.0	1.0	1.0	1.1	1.0	0.8	0.4
"SNR"	74.4	49.2	49.5	56.6	75.3	77.3	74.1	73.3	78.0	82.0	78.2
"cT"	70.4	86.2	85.6	72.7	71.7	69.8	66.1	76.8	72.3	52.6	26.8

Difference	Global	31.5	63.0	125.0	250.0	500.0	1.0k	2.0k	4.0k	8.0k	16.0k
"C50"	0.6	0.6	1.0	0.6	0.8	0.4	0.6	0.3	0.4	0.1	0.3
"C80"	0.2	0.1	0.0	0.4	0.1	0.1	0.4	0.2	0.3	0.1	0.8
"EDT"	0.1	0.3	0.1	0.2	0.1	0.1	0.1	0.1	0.1	0.1	0.2
"RT20"	0.0	0.0	0.3	0.2	0.0	0.0	0.0	0.0	0.0	0.0	0.1
"RT30"	0.0	0.3	0.5	0.1	0.0	0.0	0.0	0.0	0.0	0.0	0.2
"SNR"	27.7	19.4	17.2	9.9	24.9	26.8	28.4	28.0	30.0	33.5	16.8
"cT"	2.6	50.7	20.7	6.9	10.2	1.4	2.2	1.1	2.2	0.2	1.2

Set 4

AURORA	Global	31.5	63.0	125.0	250.0	500.0	1.0k	2.0k	4.0k	8.0k	16.0k
"C50"	−0.3	−0.7	−1.5	0.0	−2.6	−1.2	0.4	−0.8	−0.5	1.6	6.2
"C80"	2.4	2.3	2.0	2.6	0.9	1.5	3.0	1.8	2.3	4.8	10.8
"EDT"	1.2	1.4	1.1	1.1	1.3	1.2	1.1	1.3	1.2	0.8	0.4
"RT20"	1.2	1.6	1.2	1.2	1.4	1.2	1.1	1.3	1.2	0.9	0.5
"RT30"	1.2	2.8	1.3	1.2	1.5	1.3	1.1	1.3	1.2	0.9	0.5
"SNR"	49.7	35.3	38.7	41.7	51.2	50.2	48.6	51.1	49.1	49.2	64.0
"cT"	79.8	138.5	111.2	88.4	105.4	90.5	74.6	87.7	80.9	57.7	29.9

R SCRIPT	Global	31.5	63.0	125.0	250.0	500.0	1.0k	2.0k	4.0k	8.0k	16.0k
"C50"	"−1.1"	"−0.2"	"−0.2"	"−0.2"	"−1.6"	"−1.4"	"−0.4"	"−1.2"	"−0.7"	" 1.6"	" 6.5"
"C80"	" 1.9"	" 2.3"	" 2.2"	" 2.9"	" 1.3"	" 1.6"	" 2.6"	" 1.6"	" 2.1"	" 4.7"	"11.6"
"EDT"	" 1.1"	" 1.0"	" 1.0"	" 0.9"	" 1.1"	" 1.0"	" 1.0"	" 1.1"	" 1.1"	" 0.8"	" 0.4"
"RT20"	" 1.3"	" 1.5"	" 1.4"	" 1.2"	" 1.4"	" 1.2"	" 1.1"	" 1.3"	" 1.2"	" 0.9"	" 0.4"
"RT30"	" 1.3"	" 1.8"	" 1.8"	" 1.3"	" 1.5"	" 1.3"	" 1.1"	" 1.3"	" 1.2"	" 0.9"	" 0.4"
"SNR"	"78.3"	"56.0"	"56.5"	"62.9"	"72.6"	"76.7"	"76.5"	"79.5"	"78.9"	"81.9"	"77.9"
"cT"	"85.6"	"88.2"	"86.8"	"77.1"	"91.8"	"88.1"	"77.4"	"89.5"	"82.7"	"57.8"	"28.5"

Difference	Global	31.5	63.0	125.0	250.0	500.0	1.0k	2.0k	4.0k	8.0k	16.0k
"C50"	0.8	0.5	1.3	0.2	1.0	0.3	0.8	0.4	0.3	0.1	0.3
"C80"	0.5	0.0	0.2	0.4	0.4	0.2	0.4	0.2	0.3	0.1	0.8
"EDT"	0.0	0.3	0.1	0.2	0.2	0.2	0.1	0.1	0.1	0.1	0.2
"RT20"	0.0	0.1	0.2	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.1
"RT30"	0.0	0.3	0.4	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.2
"SNR"	28.7	20.7	17.9	21.2	21.5	26.5	27.9	28.4	29.8	32.8	13.9
"cT"	5.8	50.2	24.3	11.3	13.6	2.4	2.8	1.8	1.8	0.0	1.4

Set 5

AURORA	Global	31.5	63.0	125.0	250.0	500.0	1.0k	2.0k	4.0k	8.0k	16.0k
"C50"	0.7	−0.9	−0.6	0.5	−1.3	0.2	1.2	0.3	0.4	2.3	6.8
"C80"	3.6	2.2	3.0	3.2	2.8	3.5	3.7	2.9	3.4	5.7	11.6
"EDT"	1.0	1.4	1.0	1.0	0.9	1.0	1.0	1.1	1.0	0.7	0.4
"RT20"	1.0	1.5	1.1	0.9	1.0	1.0	1.0	1.1	1.0	0.8	0.4
"RT30"	1.1	5.2	1.4	1.1	1.0	1.0	1.0	1.1	1.0	0.8	0.5
"SNR"	47.5	17.9	29.7	35.0	43.7	47.8	44.7	51.4	48.0	48.5	58.3
"cT"	67.9	146.3	103.6	80.4	82.8	71.8	65.4	74.8	70.0	51.7	28.0

R SCRIPT	Global	31.5	63.0	125.0	250.0	500.0	1.0k	2.0k	4.0k	8.0k	16.0k
"C50"	0.0	−0.1	−0.1	0.3	−0.3	−0.2	0.7	−0.3	0.1	2.1	7.2
"C80"	3.3	2.4	2.3	3.1	2.9	3.5	3.5	2.5	3.1	5.5	12.2
"EDT"	0.9	1.0	1.0	0.8	0.8	0.9	0.8	1.0	0.9	0.7	0.3
"RT20"	1.0	1.5	1.5	1.1	1.0	1.0	0.9	1.1	1.0	0.8	0.4
"RT30"	1.1	1.8	1.7	1.1	1.0	1.0	1.0	1.1	1.0	0.8	0.4
"SNR"	74.4	49.2	49.5	56.6	75.3	77.3	74.1	73.3	78.0	82.0	78.2
"cT"	70.4	86.2	85.6	72.7	71.7	69.8	66.1	76.8	72.3	52.6	26.8

Difference	Global	31.5	63.0	125.0	250.0	500.0	1.0k	2.0k	4.0k	8.0k	16.0k
"C50"	0.6	0.8	0.5	0.2	1.1	0.4	0.5	0.6	0.3	0.2	0.4
"C80"	0.3	0.3	0.7	0.0	0.1	0.0	0.3	0.4	0.3	0.3	0.6
"EDT"	0.1	0.3	0.0	0.2	0.1	0.1	0.1	0.1	0.1	0.1	0.2
"RT20"	0.0	0.0	0.3	0.2	0.0	0.0	0.0	0.0	0.0	0.0	0.1
"RT30"	0.0	0.7	0.2	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.2
"SNR"	26.8	31.3	19.8	21.5	31.6	29.5	29.3	22.0	30.0	33.5	20.0
"cT"	2.5	60.1	17.9	7.7	11.1	2.0	0.7	2.0	2.3	1.0	1.2

Mean

Mean	Global	31.5	63.0	125.0	250.0	500.0	1.0k	2.0k	4.0k	8.0k	16.0k
"C50"	0.7	0.6	0.9	0.5	1.1	0.5	0.6	0.4	0.3	0.1	0.4
"C80"	0.4	0.1	0.2	0.3	0.3	0.2	0.4	0.3	0.3	0.1	0.8
"EDT"	0.0	0.3	0.0	0.2	0.1	0.2	0.1	0.1	0.1	0.1	0.2
"RT20"	0.0	0.0	0.3	0.2	0.0	0.0	0.0	0.0	0.0	0.0	0.1
"RT30"	0.0	0.3	0.4	0.1	0.0	0.0	0.0	0.0	0.0	0.0	0.2
"SNR"	27.1	23.3	16.2	17.5	24.3	26.2	27.2	26.3	28.9	32.2	19.1
"cT"	3.8	52.1	20.8	7.7	12.0	2.6	1.7	1.6	1.7	0.5	1.4

Standard Deviation

St.D.	Global	31.5	63.0	125.0	250.0	500.0	1.0k	2.0k	4.0k	8.0k	16.0k
"C50"	0.1	0.1	0.3	0.2	0.2	0.4	0.2	0.2	0.1	0.1	0.2
"C80"	0.2	0.1	0.2	0.2	0.3	0.2	0.1	0.1	0.1	0.1	0.1
"EDT"	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
"RT20"	0.0	0.0	0.1	0.1	0.0	0.0	0.0	0.0	0.0	0.0	0.0
"RT30"	0.0	0.2	0.1	0.1	0.0	0.0	0.0	0.0	0.0	0.0	0.0
"SNR"	1.3	4.7	3.1	5.1	4.3	2.4	2.0	2.6	1.6	1.7	3.9
"cT"	1.6	4.5	2.3	2.1	1.4	1.8	1.1	0.8	0.7	0.4	0.2