

SOUND SIGNAL ANALYZER

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Abstract

Analyzer system intended to visualize the sound signal in the time and frequency domain is presented. The system was realized in Matlab and it can process sound signals of wav files. Spectral analyzer takes into account the perception of the sound by humans and thus the magnitude spectrum can be visualized with frequency axis in Barks and ERBs. Spectrum can be weighted by the transfer function of A-curve.

1 Introduction

Spectral analyzers show signal spectrum with frequency abscissa in Hz. Human auditory system analyzes the signals in critical bands for which were stated units Barks and ERBs. Therefore is desired to analyze an energy of signal spectrum also in critical bands and not only in octave and one-third octave bands which are commonly used.

Presented system allows to analyze the sound signals saved in wav files. It allows to visualize signal in time domain, calculate magnitude spectrum of the sound signal and show signal spectrum like commonly used spectral analyzers and above that with frequency axis in critical bandwidth units Barks and ERBs.

2 Spectrum of a Sound Signal

Spectrum of signal $x(t)$ is defined by means of Fourier transform [1]

$$X(\omega) = \int_{-\infty}^{\infty} x(t)e^{-j\omega t} dt, \quad (1)$$

where ω is angular frequency, t is time and j is an imaginary unit ($j = \sqrt{-1}$).

2.1 Tapering of Signals

Magnitude spectrum of the harmonic signal $x(t) = \cos(\omega t)$ is composed of the two spectral components with frequencies $-\omega$ and ω . Signals in real-life processing are finite. If the spectrum is calculated from the truncated signal $x(t)$, which was truncated at places where it cannot be periodically extended, the spectrum contains additional components. This effect is called spectral leakage [2]. Signals are usually tapered by shaping window in order to reduce spectral leakage. Variety of shaping window functions exist, for example, Hamming, Bartlett and Hann [2].

2.2 RMS Value of Signals

Root Mean Square (RMS) value of the signal is defined as a value of constant which would cause the same average power as the signal. RMS value of the signal $x(t)$ is in the time domain [1]

$$x_{RMS} = \sqrt{\frac{1}{T_D} \int_0^{T_D} x(t)^2 dt}, \quad (2)$$

where T_D in case of periodic signals represents signal period or an integral number of periods and in case of non-periodical signal, it is an averaging time [1].

RMS value can be calculated in the spectral domain $X(\omega)$

$$x_{RMS} = \sqrt{\lim_{\Omega \rightarrow \infty} \frac{1}{2\Omega} \int_{-\Omega}^{\Omega} |X(\omega)|^2 d\omega}, \quad (3)$$

2.3 Octave and One-third Octave Bands

If it is talking about sound signals, the interest is usually of a spectrum in frequency range from 20 Hz to 20 kHz. Spectrum of the signal in this frequency range is usually divided into the sub-bands. The most common sub-bands used in audio analyses are octave and one-third octave bands [3].

The frequency ratio in octave is 2:1. Bandwidth of the octave bands can be calculated from signal frequency by this equation [3]

$$B = 2f - 2^{-1}f = 1.5f, \quad (4)$$

where f is a center frequency of band in Hz.

One-third octave bands are used as approximation to auditory filters which are mentioned below. Their bandwidth is defined by this equation [1]

$$B_{1/3} = 2^{1/6}f - 2^{-1/6}f = 0.232f, \quad (5)$$

where f is a center frequency of band in Hz.

2.4 Critical Bands

Sound signal enters the ear and is delivered to the inner ear where is a cochlea. Inner hair cells are placed along the basilar membrane situated in the cochlea. They work as receptors, convert mechanical vibrations of basilar membrane into the neural signal. The basilar membrane response is tuned and thus the entire auditory system is tuned. The function of basilar membrane can be roughly likened with the function of auditory filterbank. Critical bandwidth is the bandwidth of auditory filters [1].

Critical bandwidth was determined over the years. Bark scale was published by Zwicker in 1961. Different set of critical bandwidths was measured in Cambridge, England in the 1980, it is called ERB (Equivalent Rectangular Bandwidth) scale. Approximate expressions for bandwidths of Bark and ERB critical bands are [1]

$$B_{Bark} = 25 + 75(1 + 1.4F^2)^{0.69}, \quad (6)$$

and

$$B_{ERB} = 24.7(1 + 4.37F), \quad (7)$$

where F is the center frequency of the band in kHz. The Fig. 1 shows bandwidth of octave, one-third octave, Bark and ERB bands as a function of frequency. The table 1 shows center frequencies sub-bands used in the spectral analyzer presented in this paper.

Frequency scale can be transformed to an absolute scale defined by a critical-band number. The critical band number represents the number of critical bands needed to cover certain region

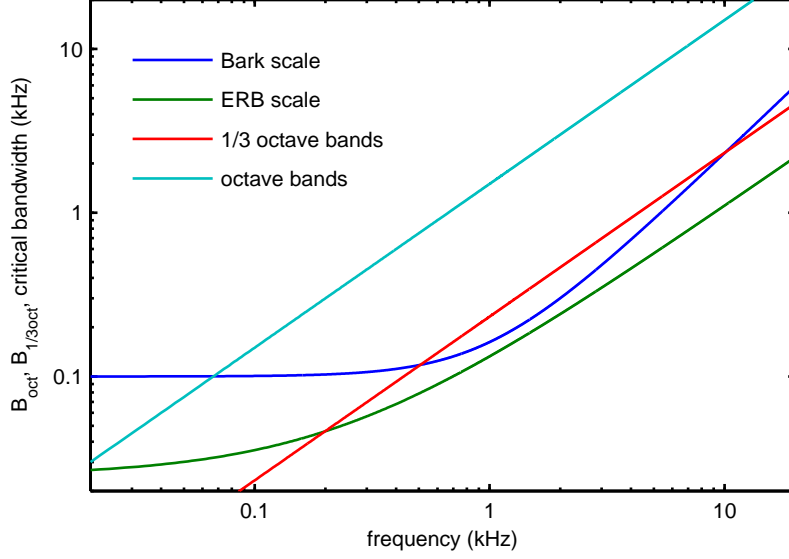


Figure 1: The critical bandwidth in Barks scale and ERBs. The bandwidth of octave and one-third octave bands.

of a frequency axis if they are stacked consecutively. Critical band number is represented by the symbol z .

Approximation of bark scale is done by these equations [1]

$$z'(f) = \frac{26.81f}{1960 + f} - 0.53 \quad (8)$$

$$z = \begin{cases} z' + 0.15(2.0 - z') & \text{if } z' < 2.0 \\ z' + 0.22(z' - 20.1) & \text{if } z' > 20.1 \\ z' & \text{otherwise} \end{cases} \quad (9)$$

where f is frequency in Hz.

Transformation of frequency in Hz to ERB scale is done by this formula [1]

$$z(f) = 21.3 \log(0.00437f + 1), \quad (10)$$

where f is frequency in Hz.

Bark and ERB scales as a function of frequency are shown in the Fig. 2.

Octave bands center frequencies
16, 31.5, 63, 125, 250, 500, 1000, 2000, 4000, 8000, 16000 Hz
One-third octave bands center frequencies
16, 20, 25, 31.5, 40, 50, 63, 80, 100, 125, 160, 200, 250, 315, 400, 500, 630, 800, 1000, 1250, 1600, 2000, 2500, 3150, 4000, 5000, 6300, 8000, 10000, 12220, 16000, 20000
Center frequencies of the critical bands in Barks
50, 150, 250, 350, 455, 570, 700, 845, 1000, 1175, 1375, 1600, 1860, 2160, 2510, 2925, 3425, 4050, 4850, 5850, 7050, 8600, 10750, 13750
Center frequencies of the critical bands in ERBs
30, 61, 94, 131, 172, 218, 269, 326, 389, 459, 538, 625, 723, 831, 952, 1086, 1236, 1403, 1589, 1796, 2027, 2284, 2570, 2889, 3244, 3640, 4080, 4571, 5118, 5727, 6406, 7161, 8003, 8941, 9986, 11149, 12446, 13890, 15498, 17290, 19285

Table 1: Center frequencies of sub-bands used in the presented spectral analyzer.

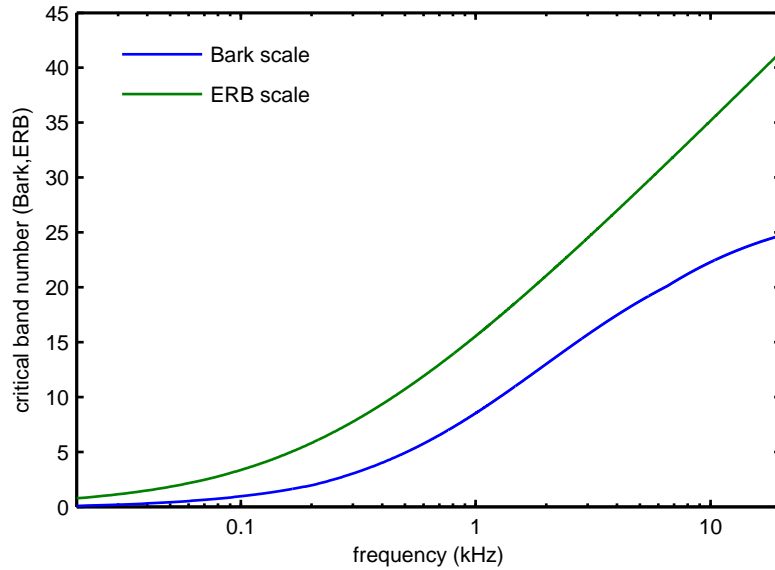


Figure 2: Conversion from frequency in Hz to Bark and ERB scales.

2.5 Weighting of Signal Spectrum

Human auditory system perceived different spectral components of the sound signal with different loudness. The auditory system is the most sensitive in the frequency region between 1 and 3 kHz. Sensitivity then decrease towards lower frequencies (deep tones) and higher frequencies (high pitch) [3]. This effect is modeled by weighting the sound signal spectrum by a filter function attenuating lower and higher frequency regions. The most common used filter function is called A-curve. Transfer function of A-curve filter is shown in the Fig. 3

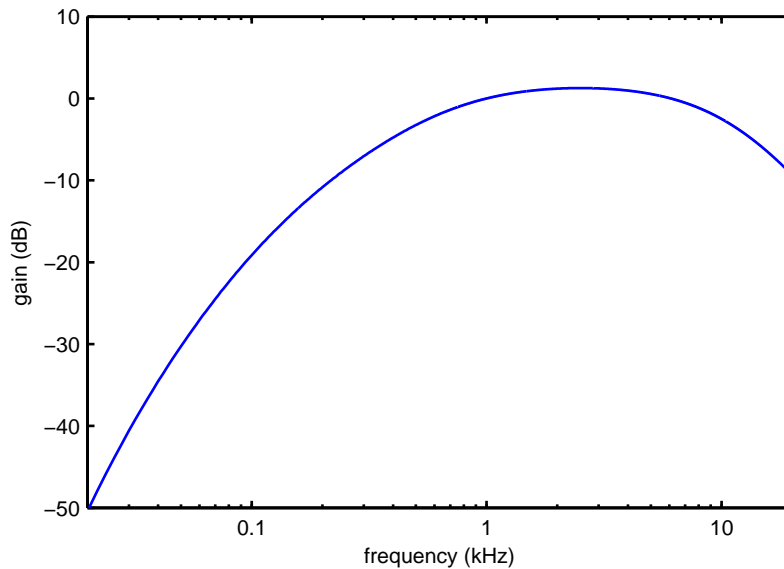


Figure 3: Weighting A-curve

3 Sound Signal Analyzing System

System was realized in Matlab with graphical user interface. Main window of the system (Fig. 4) offer you a possibility to activate an oscilloscope and spectral analyzer. Each system is opened

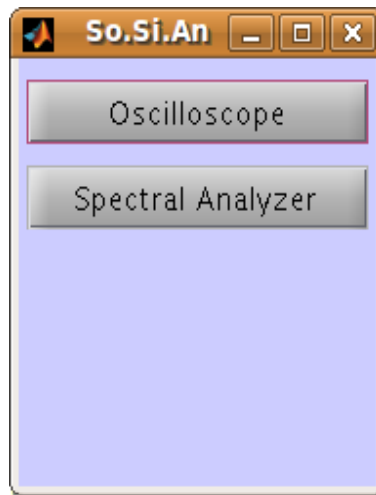


Figure 4: Main window of the Sound Signal Analyzer

in separate figure. The spectral analyzer can be also activated from the oscilloscope. System is able to load sound signals in wav files.

The realisation was done in the graphical user interface. The main script launch the main window shown in the Fig. 4. Oscilloscope and Spectral Analyzer are implemented as a callback functions. System was not realized in graphical user interface development environment (GUIDE), all properties of the used control elements were set directly in the Matlab editor.

3.1 Oscilloscope

The main window of the oscilloscope can be seen in the Fig. 5. Wav files load button and table with the informations about loaded signal are in the upper part of the figure. Timebase and

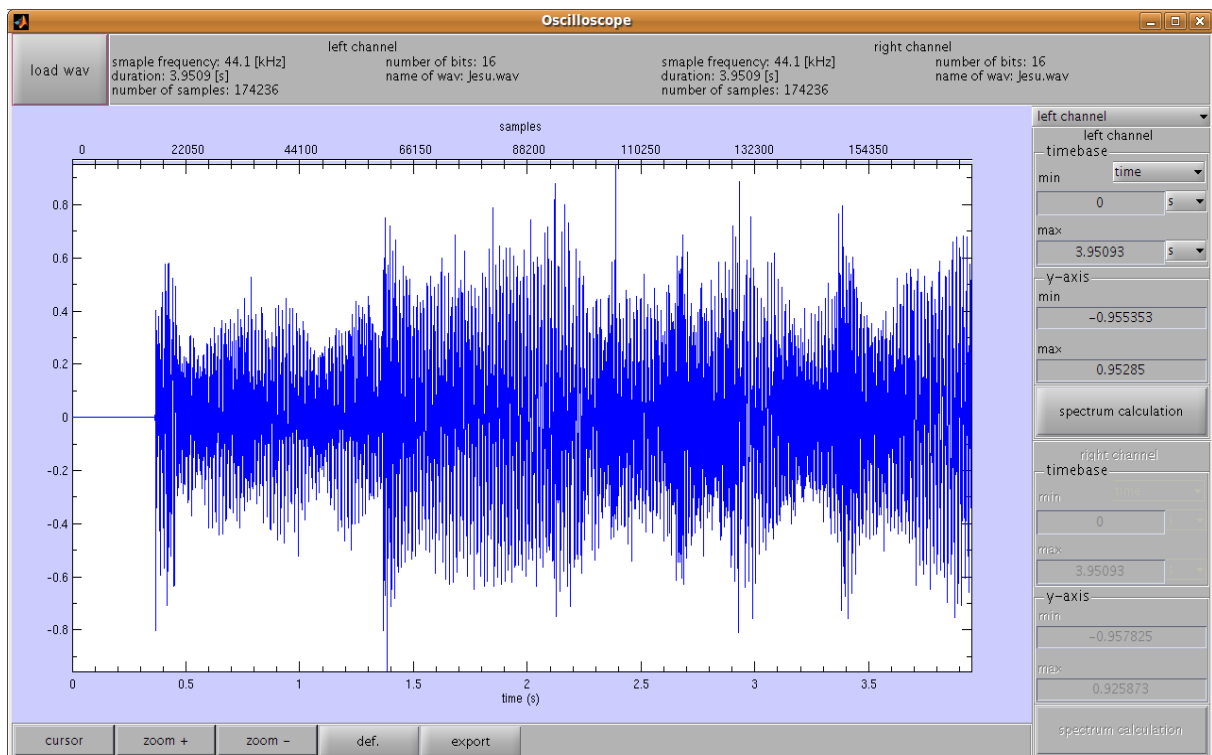


Figure 5: Oscilloscope window

y-axis control elements are situated in the right side of the figure. Bottom part of the figure contains menu allowing to zoom visualized signal, to export it into the separate figure, cursor button and button with default settings of the figure and plot axis.

System is able to load mono and stereo wav files. It is possible to load only one channel of the wav file and to choose to which of the oscilloscope channel it will be placed. Oscilloscope has two channels, left and right. System allows to visualize the loaded signal in main four regimes. It is left channel, right channel, left and right channel in different graphs and left and right channel in the same graph. Each graph has two x-axis, bottom axis is time and upper axis is samples of the signal.

Very useful function of the oscilloscope is the possibility to launch the spectral analyzer. It is done by pushing the button at the bottom part of the control menu called Spectrum calculation. Visible part of the signal is then used in the spectral analyzer. Signal is loaded into the same channel of the spectral analyzer as it was in the oscilloscope.

3.2 Spectral Analyzer

The spectral analyzer figure has same positioning of the information and control elements as the aforementioned oscilloscope figure. It has also two channels, left and right. Stereo or mono wav files can be loaded into the analyzer and their spectrum is then calculated and visualized in the graph. System offers the possibility to visualize magnitude spectrum in four main regimes, left channel, right channel, left and right channel into the different graphs and left and right channel into the same graph.

Upper x-axis in the graphs is samples of signal spectrum and lower axis is frequency. It is possible to choose different units of frequency axis (Hz, Barks or ERBs) and to show frequency axis linearly or logarithmically. Y-axis should be shown in dB or linearly in absolute values. Spectrum can be also weighted by A-curve filter function to model the frequency dependence of loudness perception. Very important function of the analyzer is a tapering of the signal by

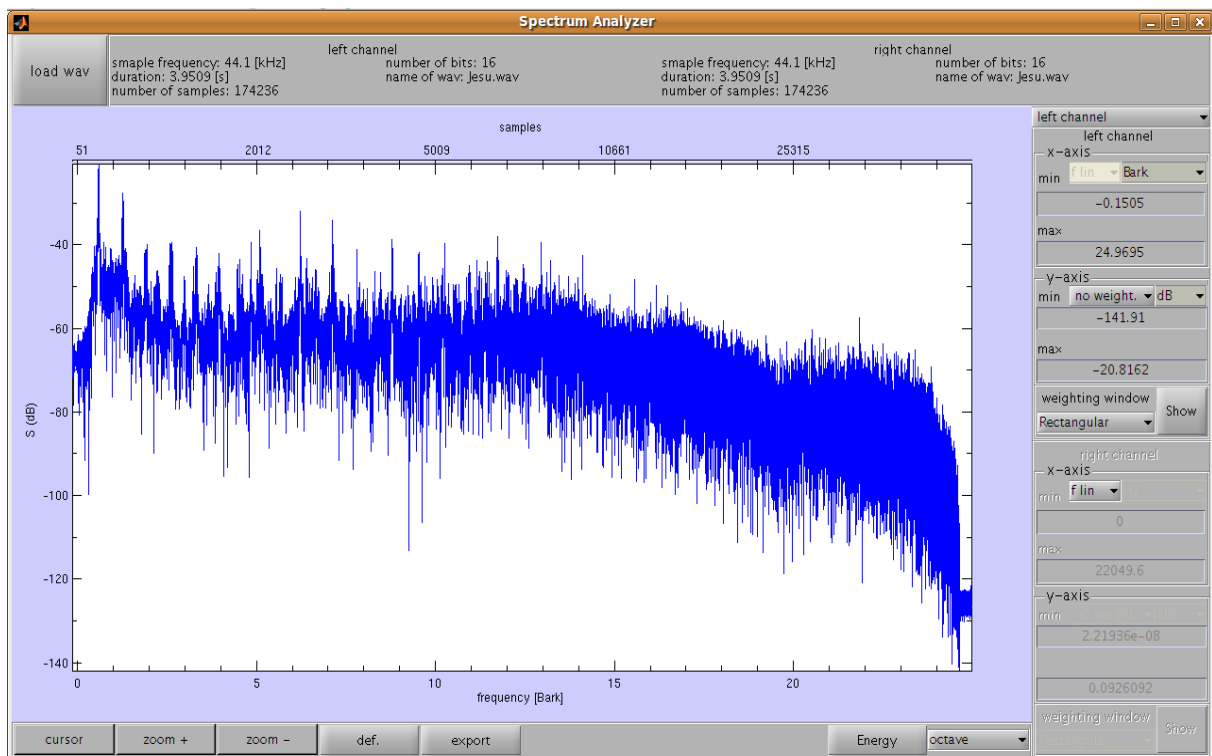


Figure 6: Spectral Analyzer

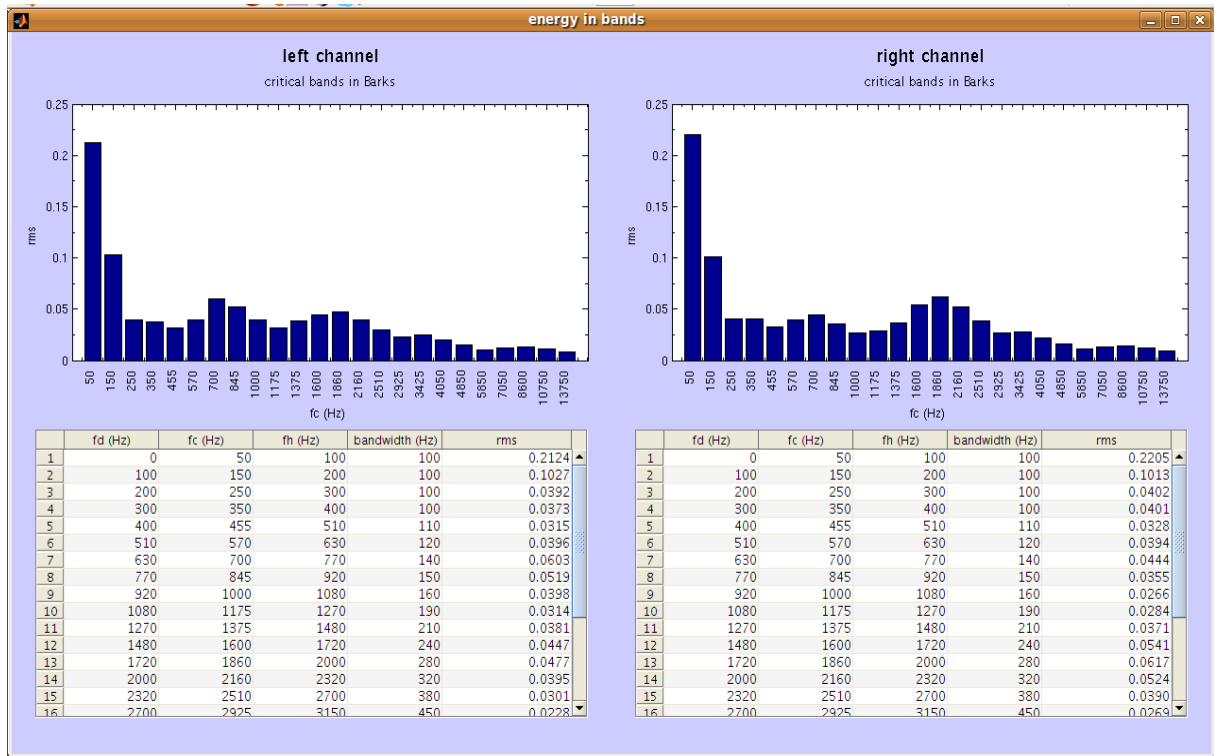


Figure 7: Root mean square values in sub-bands

window function in order to reduce the spectral leakage effect. The system allows to choose one of the variety of tapering window which are implemented in the Signal Processing Toolbox.

Spectrum is calculated by means of fast Fourier transformation (fft function in Matlab). The fft function calculates spectral coefficients divided between $-fs/2$ and $fs/2$, fs is a sample frequency of wav file. Presented spectral analyzer shows only the part of magnitude spectrum from 0 to $fs/2$ and thus the spectrum needs to be scaled in order to reach equality between signal root mean square value calculated in the time and frequency domain.

The button called energy which is situated on the bottom part of the figure serves to root mean square values calculation in sub-bands of the signal spectrum. Root mean square values can be calculated in the octave, one-third octave, Bark and ERB bands. Energy is calculated using the equation 3 and displayed in the separate figure. Data are also displayed in the bar graph as is shown in the Fig. 7.

4 Conclusion

Sound signal analyzing system implemented in Matlab was presented in this paper. System allows to analyze the sound signals in the time domain (oscilloscope) and frequency domain (spectral analyzer). Oscilloscope and spectral analyzer can be launched and work independently. System is able to process the sound signals in wav files.

Oscilloscope and spectral analyzer were designed in order to reach the most similarity with hardware oscilloscopes and analyzers. System is intended to analyze sound signals and thus spectral analyzer allows to visualize a signal spectrum on frequency axis in units related with critical bands, Barks and ERBs. It is also possible to calculate root mean square value of the signal in octave, one-third octave, Bark and ERB bands. System will serve at the department of radio engineering, FEE, CTU in Prague for research and teaching.

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