

# Loudness Measurement according to EBU-R128

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- Radio listeners and TV viewer hate it when they have to adjust the volume all the time. Why does this problem exist ?
  - \* The nature of contemporary broadcast content.
  - \* Decline of technical standards due to commercial pressure.
  - \* Automated play-out.
  - \* 'Out of context' production workflow.
  - \* The *Loudness Wars*.
- Broadcasters are aware of the problem.



- Automated loudness measurement that
  - \* does not require human interpretation,
  - \* can be applied to stored data before it is used,
  - \* produces reliable results over e.g. a complete song, or a complete program.
  
- Technical standards.
  
- Consumer pressure and legislation.



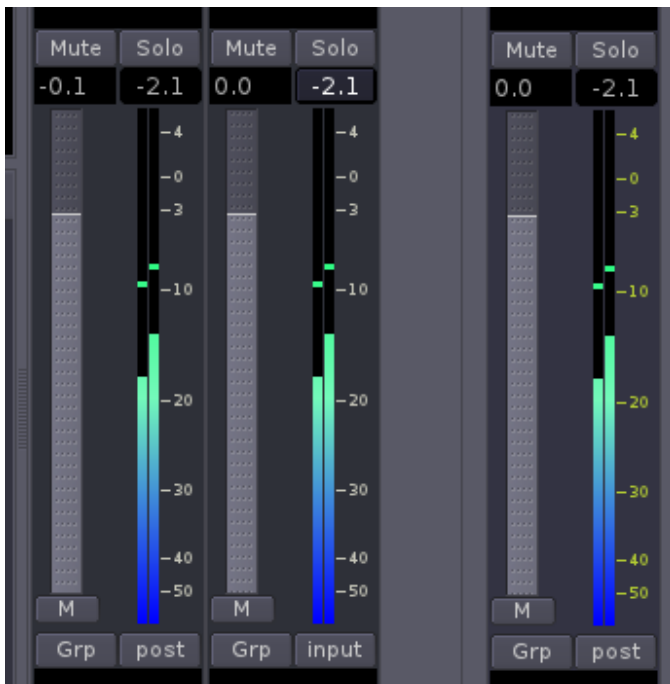
- A better listener and viewer experience.
- Maybe a solution for the Loudness Wars:

*If broadcasters and consumer playback devices can measure loudness before playing a song and apply an automatic level correction, then there is nothing to be gained from pushing up recording levels at the expense of quality.*



- Very few professional tools seem to exist (e.g. Dolby LM100).
- The film industry seems to have its act together.
- The music industry, and radio and TV broadcasters absolutely not.
- Current level measurement practices do not provide a solution.

# Levels meters: Digital peak meter



- You all know these...
- Found in nearly all audio software.
- Measure peak sample value with slow fallback.
- Essential to check digital recording levels.
- **No useful loudness indication at all.**

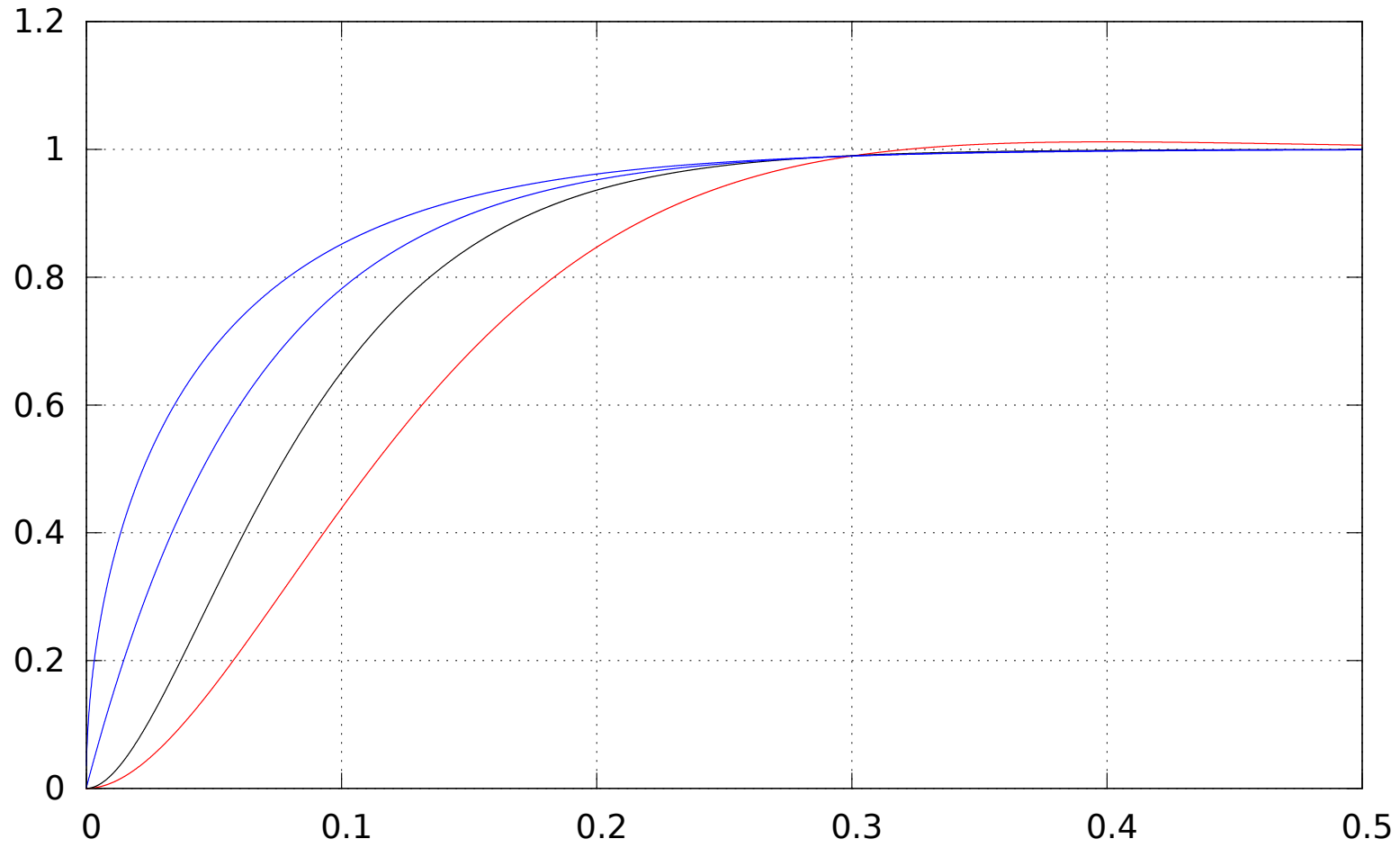


- BBC form shown, many others exist.
- Widely used in broadcasting (at least in Europe).
- Measure average level with fast rise time (10ms) and slow fall-back.
- Defined by international and corporate standards.
- Provide some indication of loudness but requires human interpretation.

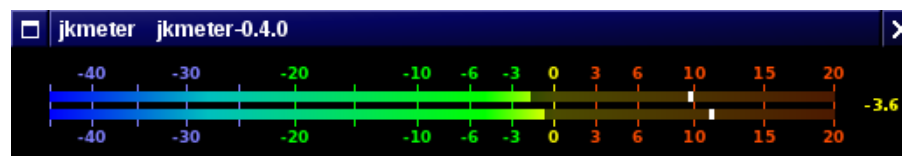


- Popular in broadcasting and music studios in the U.S.A.
- Measures average of absolute value.
- Dynamic behaviour defined by the mechanics of the meter.
- Provide some indication of loudness but not very accurate.
- Standard form has limited dynamic range.
- Most software implementations get it wrong.





- RMS, 1st order filter
- Average, 1st order filter
- Average, 2nd order, no overshoot
- Correct response



- Designed by mastering expert Bob Katz.
- Measures RMS and digital peak, displayed on the same scale.
- 'OdB' for RMS measurement offset by 20 or 14 dB.
- Not widely used, but popularity is rising.
- No official standards.
- Provides *quite a good indication of loudness*.



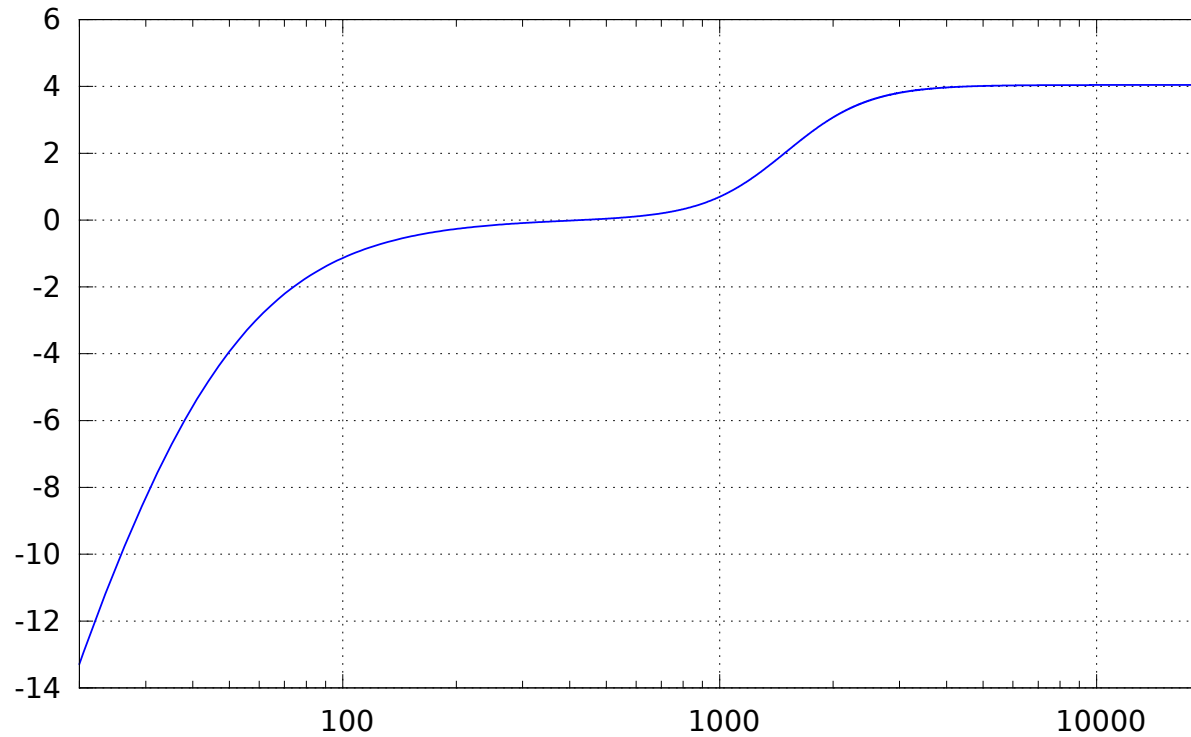
- With the exception of the K-meter, none of the currently used level meters provide a reliable loudness indication.
- VU and PMM require human interpretation and skilled users.
- They only provide *momentary* values, there is no standard for *integrated* measurement, nor for the determination of *loudness range*.



- Under the impulse of Florian Camerer (senior sound engineer at the ORF) the European Broadcasting Union has taken the initiative to define a loudness measurement standard.
- The *PLOUD* Working Group has produced Recommendation R-128, which is in turn based on standards defined by the ITU.
- R-128 defines methods to measure *integrated loudness* and *loudness range*, and some standards on how these values should be displayed.
- Implementation guidelines and audio test files are provided as well.
- Commercial equipment and software based on R-128 is starting to become available. Linux Audio should follow !

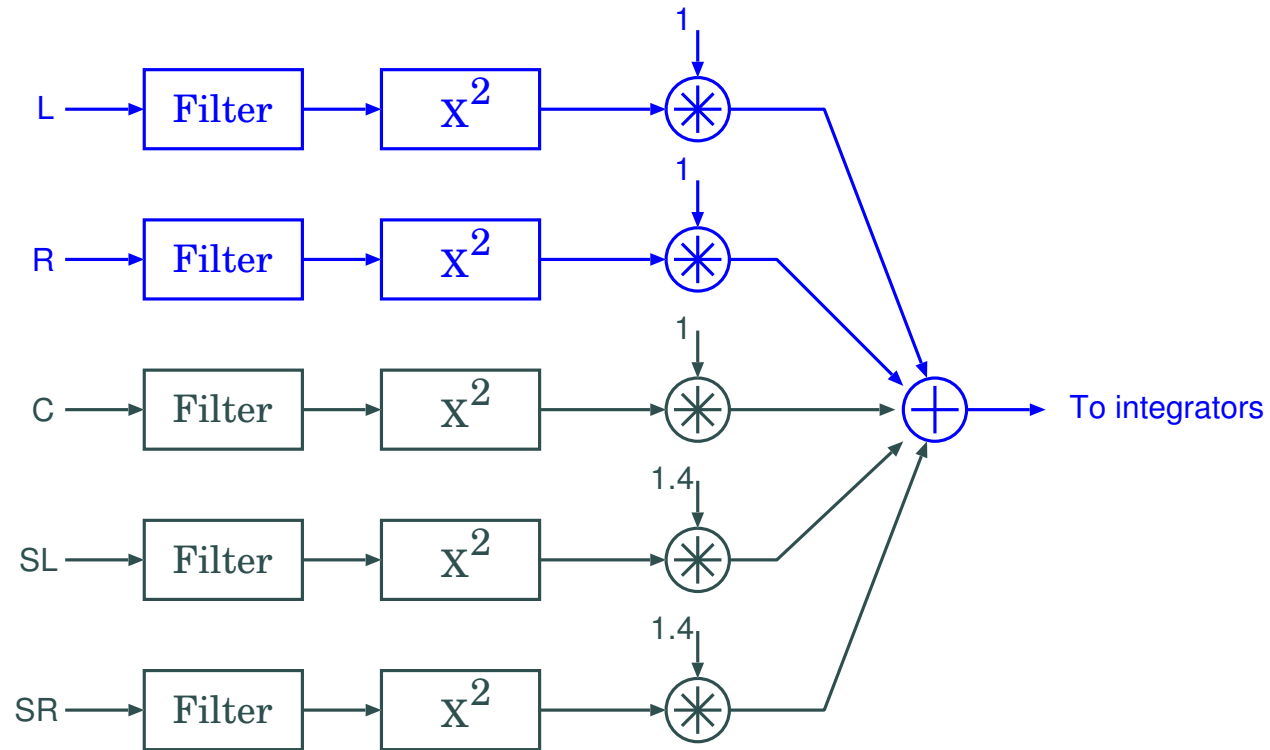


- R-BS1770 defines the basic loudness measurement algorithm. It is based on years of research by various institutions and members of the ITU.
- The algorithm has been validated by extensive listening tests which have shown very good correlation between measured and subjective results.
- Key features of the algorithm are:
  - \* Pre-detection filtering.
  - \* RMS measurement on individual channels.
  - \* Channel *powers* are summed.
  - \* Per-channel weights for 5.1 surround.
- The ITU recommendation does not specify a reference level.



- Combination of second order highpass and second order shelf.
- Defined as a combination of two biquad sections.
- Gain needs to be normalised at 1kHz.

# The ITU algorithm





- R-128 builds on the ITU recommendation and defines the the algorithms to compute four outputs:
  - **M** : momentary loudness
  - **S** : short term loudness
  - **I** : integrated loudness
  - **LRA** : loudness range
- It also specifies the reference level as **-23 dB** w.r.t. a sine wave at maximum digital level.
- Measured values should be displayed either as *absolute*, using the unit **LUFS**, or relative to the reference level and using the unit **LU**. For both, this unit has the same meaning as the dB.
- R-128 also defines the ranges and update rates for the display, some required controls and annotation, but not e.g. the exact layout or colors.





- The **M** output is computed by integrating the sum of powers over a sliding rectangular window of 400 ms.
- The **S** output is computed by integrating the sum of powers over a sliding rectangular window of 3 seconds.
- An instrument or software application conforming to R-128 should allow the user to display either **M** or **S**. Update rate must be at least 10 times per second.
- The maximum values for either must be displayed as well.
- For a graphical display two ranges should be provided: one from -18 to +9 LU, the second from -36 to +18 LU.
- The user should be able to select either the relative (LU) scale, or the absolute (LUFS) one.



- The **I** algorithm provides a loudness value averaged over an arbitrary long time interval.
- It can be applied to either an audio file, or in interactive mode, controlled by *start*, *stop* and *reset* commands. The same controls also enable and reset the display of the maximum values of **M** or **S**.
- The measurement is based on the 400 ms windows used for the **M** display. The integration periods must overlap by at least 200 ms.
- Given this input, the integrated loudness is computed in four steps:
  - \* All inputs below -70 dB (re. full scale) are discarded.
  - \* The average power of the remaining values is computed.
  - \* All inputs lower than 8 dB below this average are discarded.
  - \* The output is the average power of the remaining values.



- The **LRA** algorithm provides an indication of the *loudness range* over an arbitrary long time interval. It can be used e.g. to determine if some compression is required.
- It can be applied to either an audio file, or in interactive mode, using the same controls as for the integrated measurement.
- The algorithm is designed to ignore periods of almost silence, and very loud but short sounds (e.g. a gunshot in a movie).
- R-128 does not require endpoints of the range to be displayed, only their difference.
- The measurement is based on the 3 second windows used for the **S** display. The integration periods must overlap by at least 2 seconds.

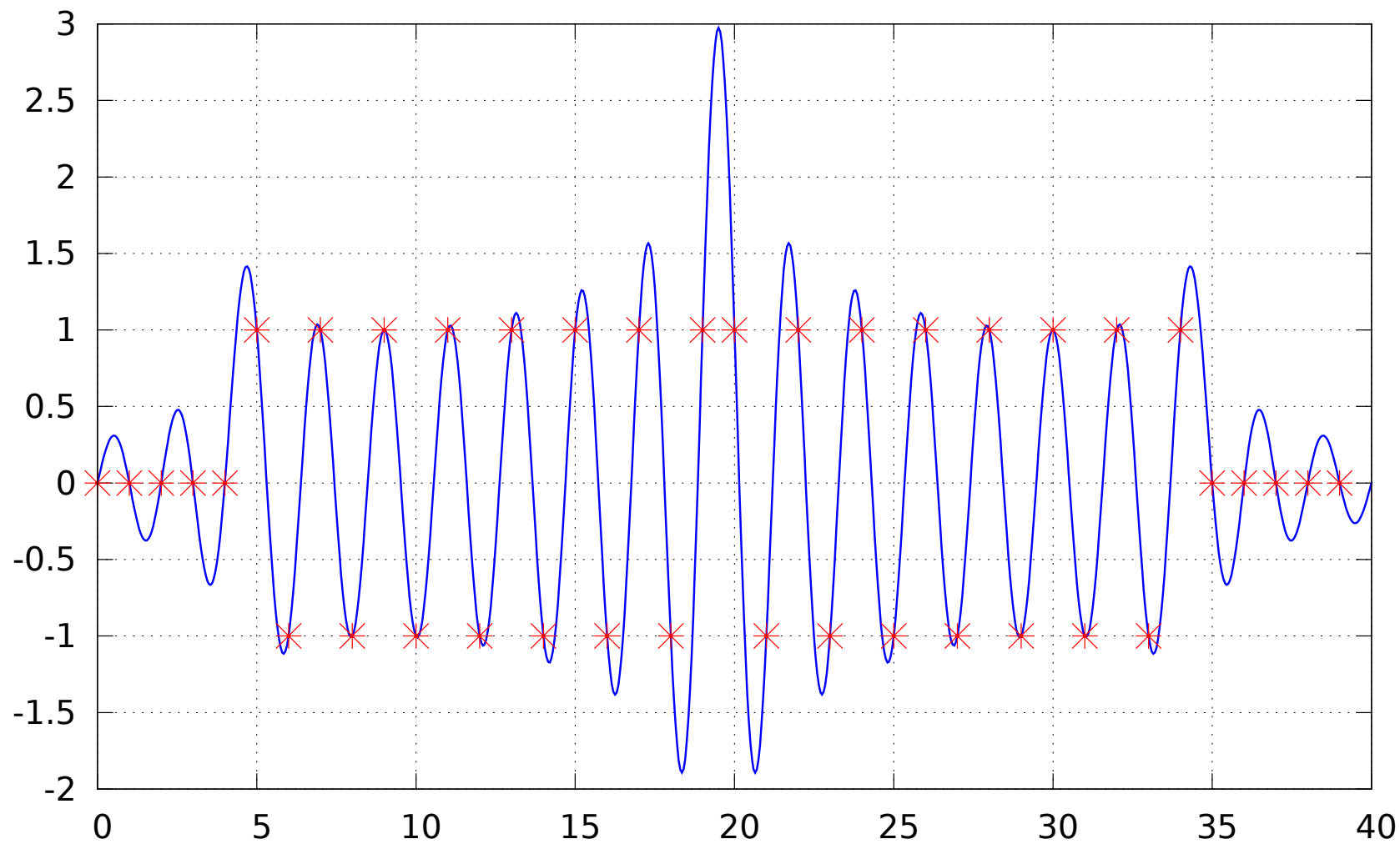


- Given the values used for the **S** measurement, the loudness range is calculated in four steps:
  - \* All values below -70 dB (re. full scale) are discarded.
  - \* The average power of the remaining inputs is computed.
  - \* All inputs lower than 20 dB below this average are discarded.
  - \* The loudness range is then computed as the difference between the level that is exceeded by 90% of the remaining values, and the one exceeded by 5% of them.
- The use of *percentiles* to determine the final output provides for most 'robust' estimation.



- EBU-R128 also requires an indication of peak value going above limits.
- Inter-sample values easily exceed sample values.
- The ITU documents suggest to upsample by at least a factor of four to find the real peak values.

# Inter-sample peaks

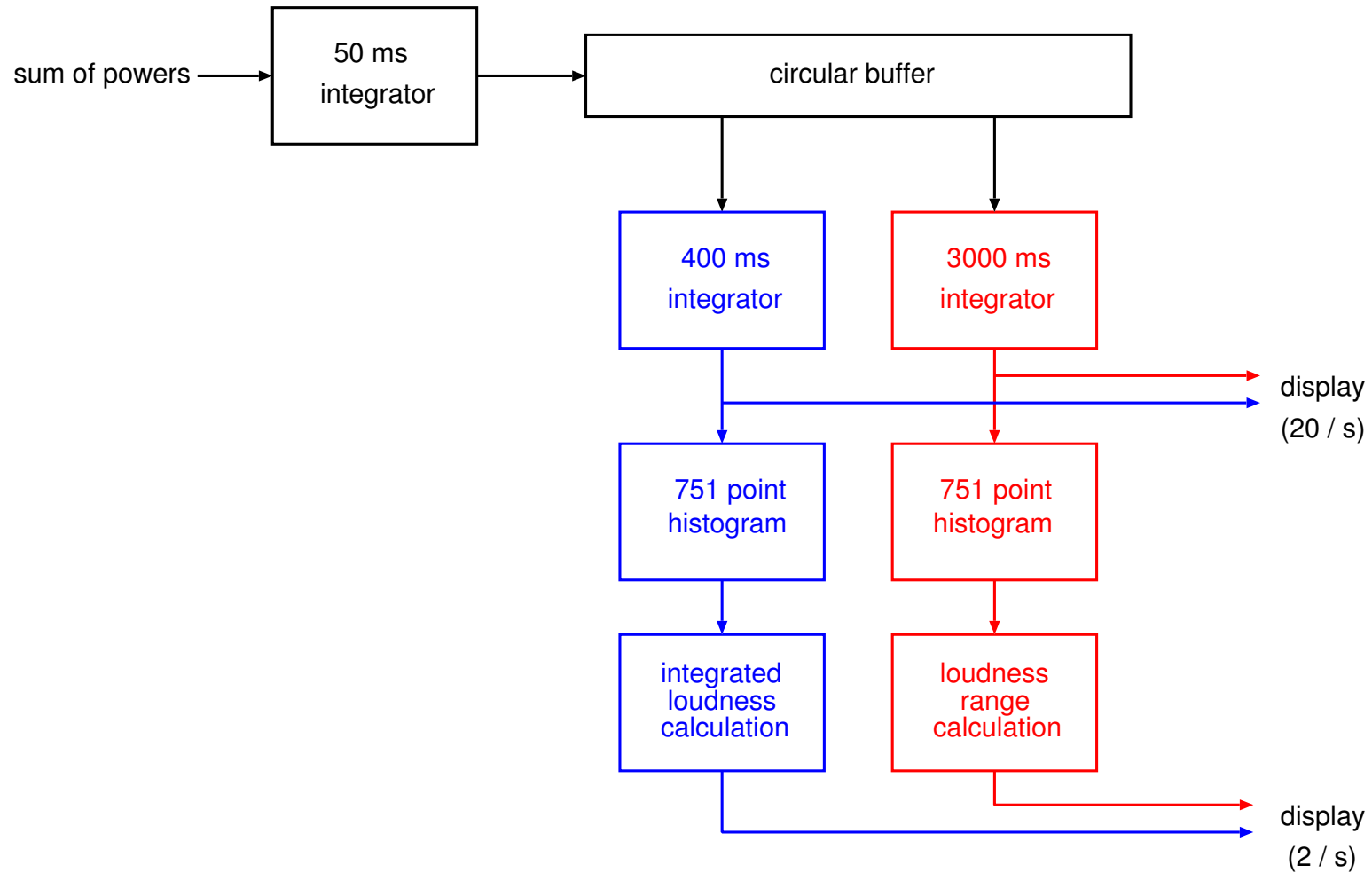


— original samples    — interpolated waveform



- A naive implementation (and the Matlab code provided by the EBU) would require unbounded storage, and the complexity of the calculation would increase as the time interval becomes longer.
- Both problems can be avoided by the use of *histogram* data structures:
  - \* Fixed data size for any length of the history.
  - \* Efficient fixed-time calculation of averages and percentiles.
- The implementation presented uses two histograms with 751 'bins' each, covering the range -70 to +5 dB with a step of 0.1 dB. A small step size removes the need for interpolation in the histogram data.
- The input data rate for the **I** and **LRA** algorithms is twice the minimum required by R-128.

# Implementation of the EBU-R128 algorithms - 2



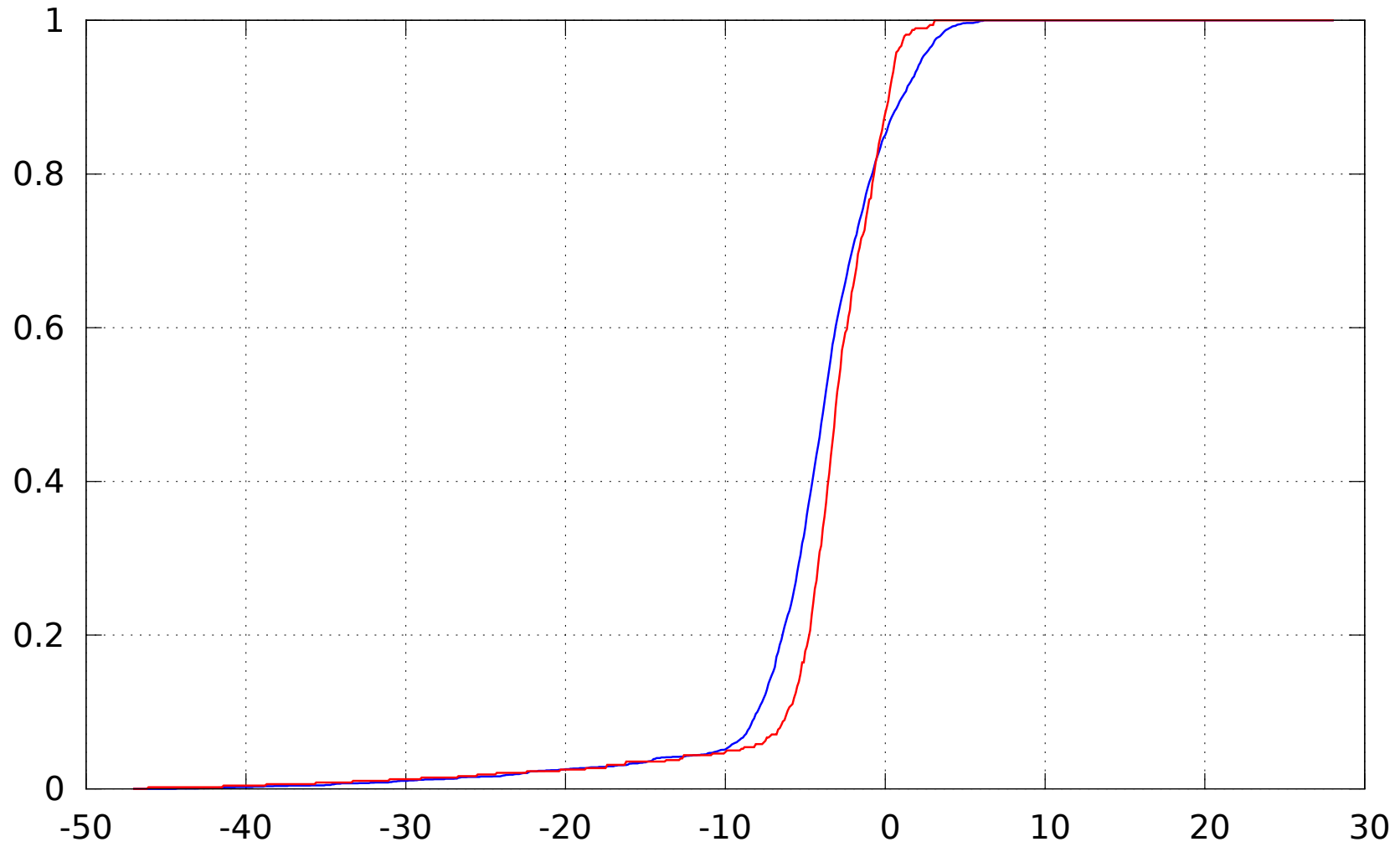




- Two applications are provided in the first release:
- **ebumeter**
  - \* Real-time measurement with GUI.
  - \* First release does not have peak indicators.
  - \* Future version may include graphical display of loudness history.
- **ebur128**
  - \* Command line app to measure audio files.
  - \* Provides some extra information (internal values).
  - \* Optionally produces a cumulative probability data file that can be displayed using Gnuplot.

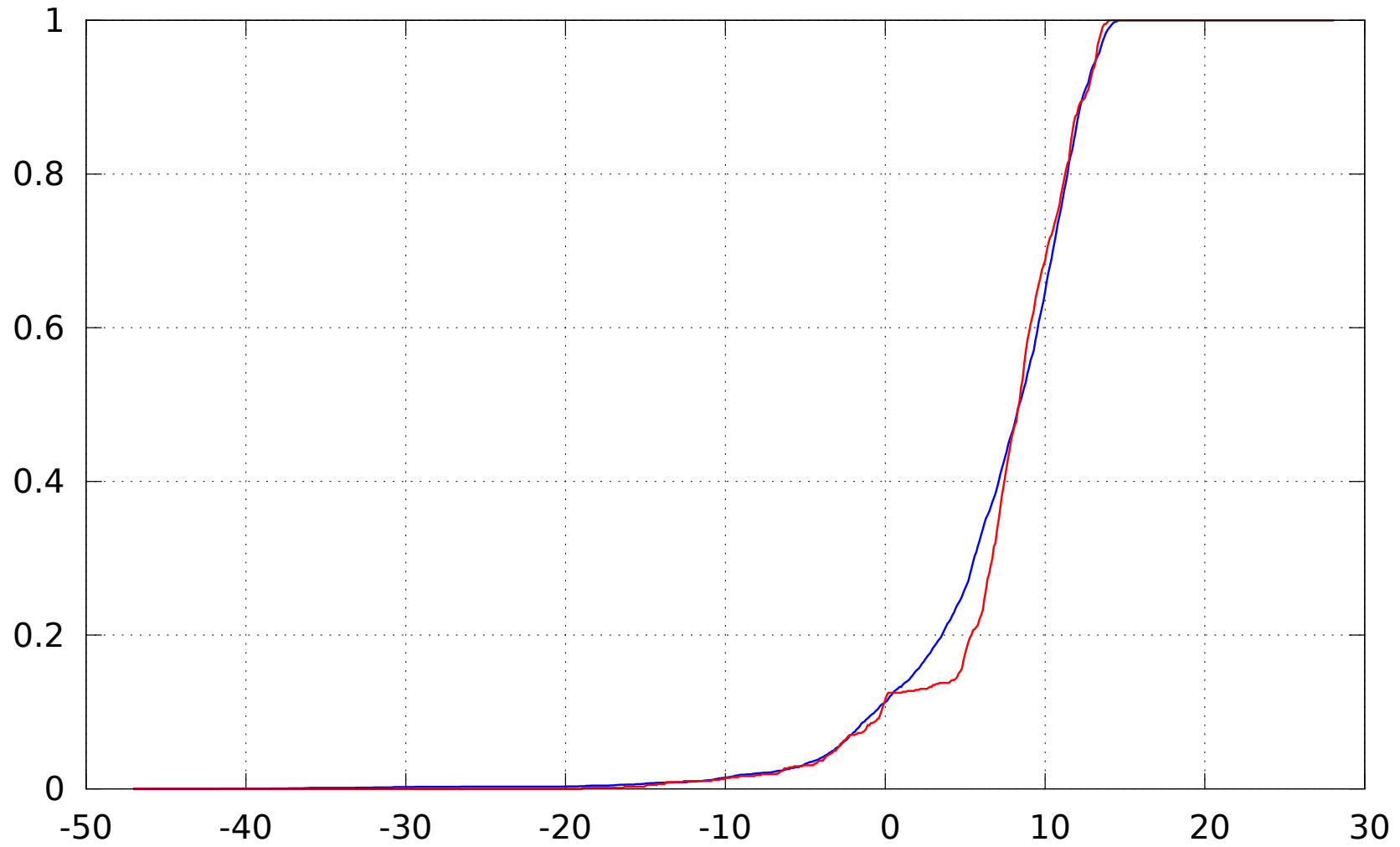


Title	I	LRA	pk	pk-I
Sting - Moon over Bourbon street	-2.1	6.3	-8.2	16.9
Weather Report - A remark you made	+6.1	11.1	-1.1	15.8
D. Fagen - Maxine	+2.0	7.1	-3.6	17.4
Pink Floyd - Money	+9.8	13.3	-0.4	12.8
Sheryl Crow - All I wanna do	+7.4	2.7	0.0	15.6
F. Mendelssohn - Die Hebriden	+4.0	19.9	-0.3	18.7
D. Shostakovich String Quartet 3	+6.0	14.0	-1.1	15.9
B. Britten - Sea Interludes	+4.0	18.7	-0.1	18.9
A. Bruckner - Symphony 9 - 1	+6.2	24.4	0.0	16.8
A. Bruckner - Symphony 9 - 2	+6.9	23.8	0.0	16.1



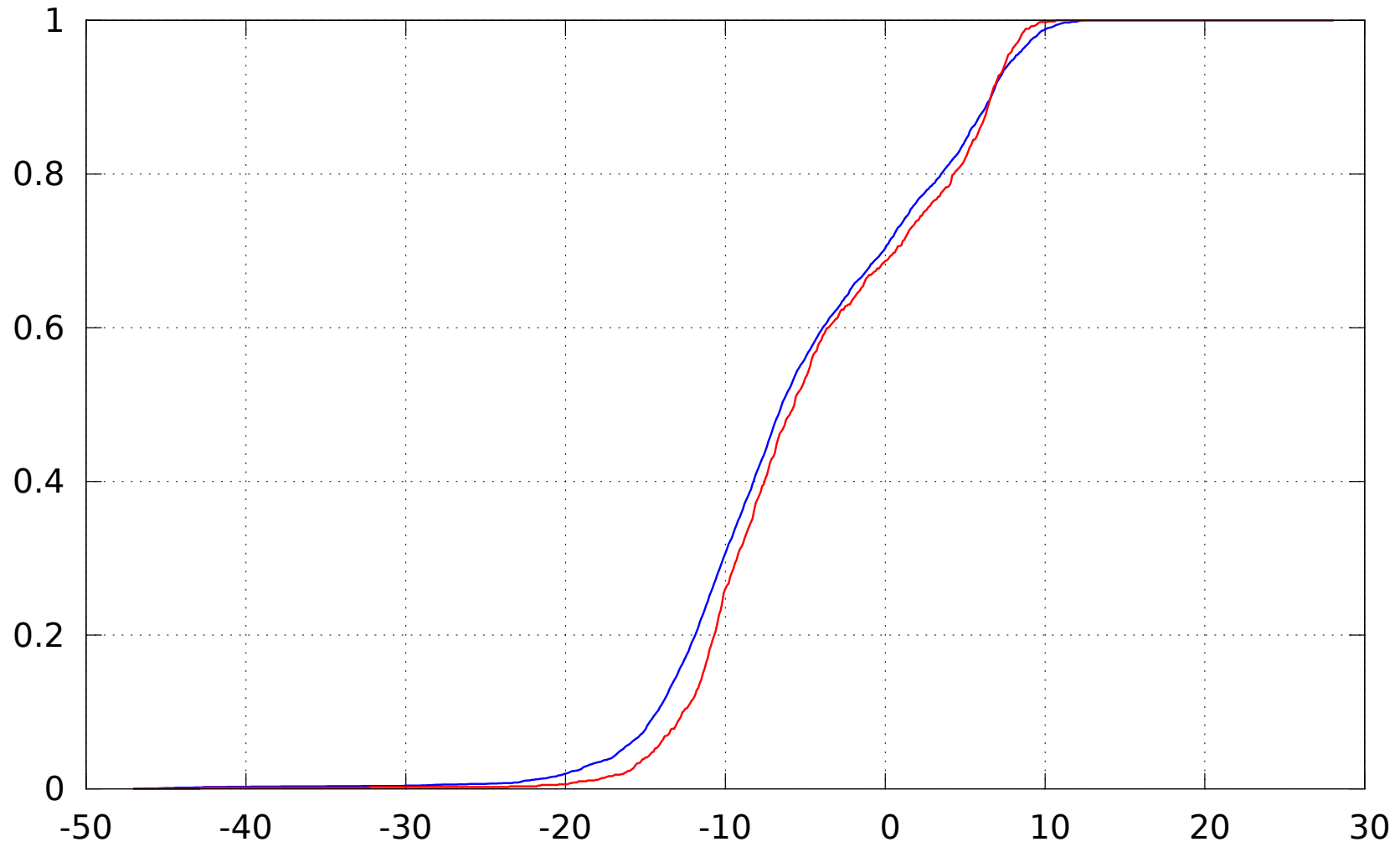
*Sting – Moon over Bourbon street*

■ 400ms window ■ 3s window



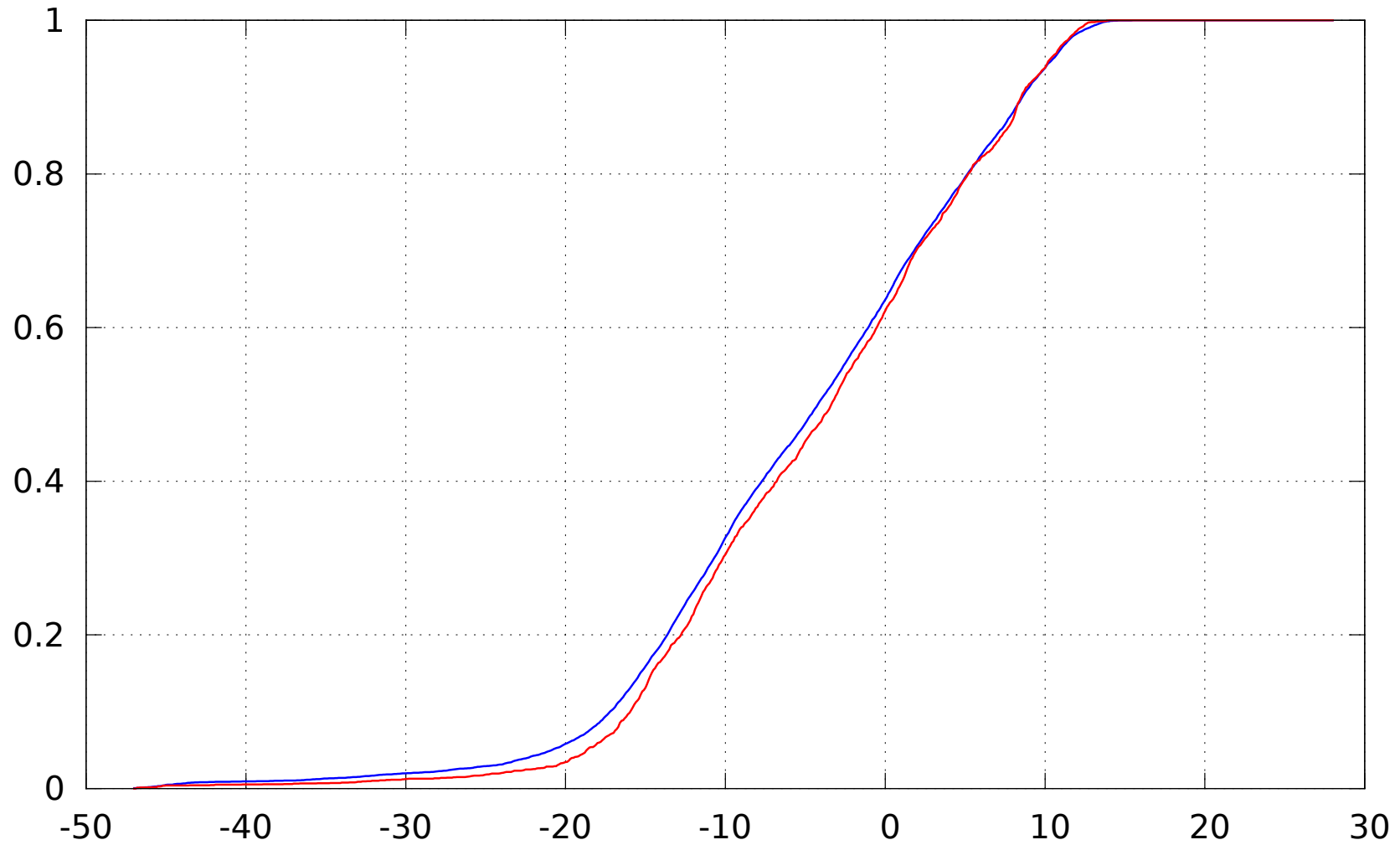
*Pink Floyd - Money*

■ 400ms window ■ 3s window



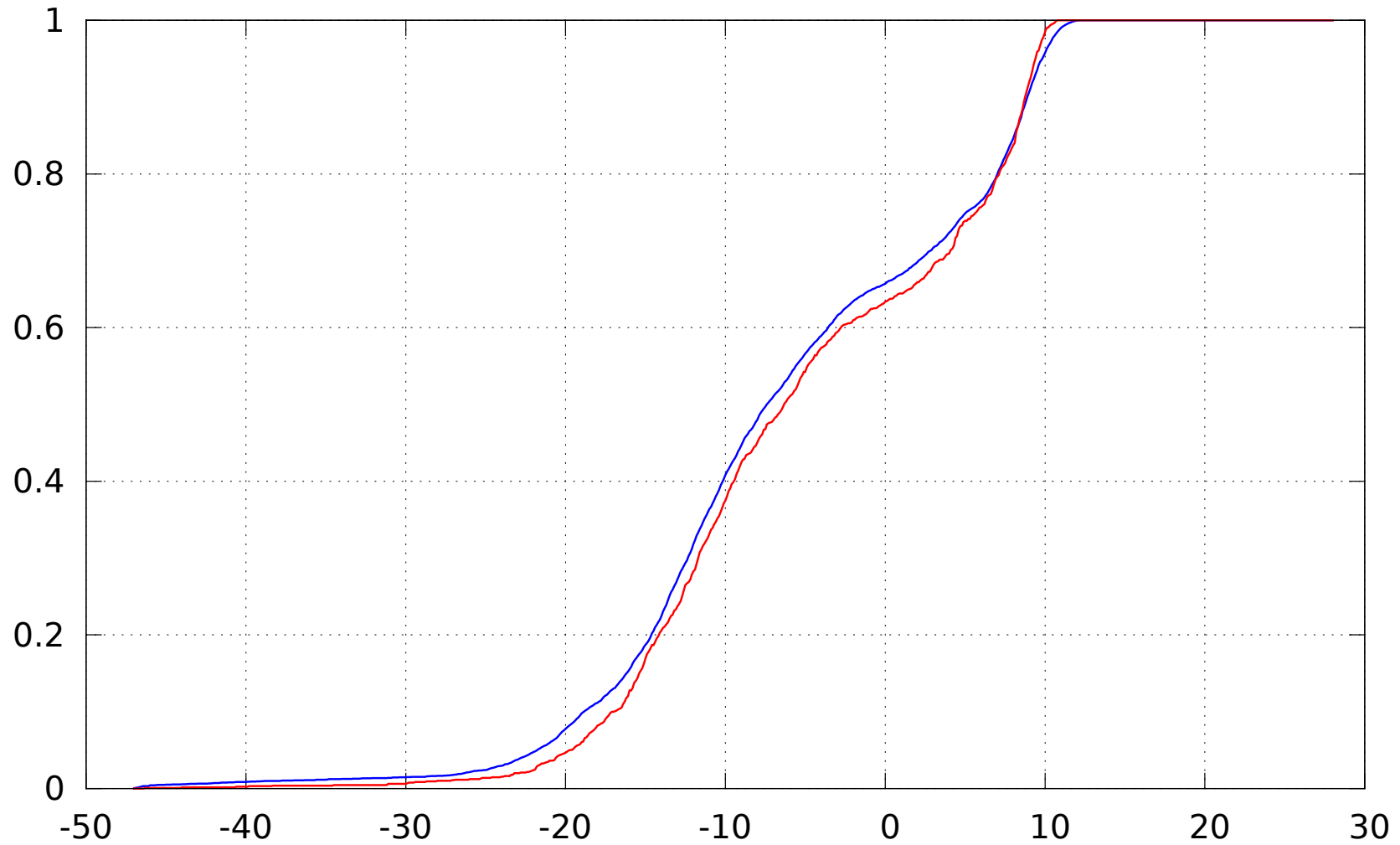
*F. Mendelssohn – Die Hebriden*

■ 400ms window ■ 3s window



*A. Bruckner – Symphony No. 9, Feierlich, misterioso*

— 400ms window — 3s window



A. Bruckner – Symphony No. 9, Bewegt, lebhaft

— 400ms window — 3s window



The end