CLAIMS

WHAT IS CLAIMED IS:

1. A method of audio signal processing, the method comprising:

receiving a data block of an audio signal comprising a first number of samples;

for each sample of the first number of samples, slicing a portion of the audio signal corresponding to the sample into a particular number of sub-samples;

obtaining a total number of sub-samples within the data block based on a multiplication of the first number with the particular number;

resampling the data block of the audio signal into a second number of samples that is equal to the first number plus one, when an underrun condition occurs in an electroacoustic device, or to the first number minus one, when an overrun condition occurs in the electroacoustic device;

distributing the second number of samples evenly within the total number of sub-samples obtained such that a duration corresponding to the second number of samples is equal to a duration corresponding to the first number of samples, the first number of samples and the second number of samples having been derived from the same data block; and

playing back the resampled data block of the audio signal via the electroacoustic device.

2. The method of claim 1, wherein the particular number is a power of 2.

3. The method of claim 2, wherein arithmetic shift operations of an exponent number of bits of the data block are used instead of division operations in the resampling of the data block.

4. The method of claim 1, wherein the portion of the audio signal corresponding to the sample is a portion of the audio signal between the sample and a next sample immediately subsequent to the sample.

5. The method of claim 4, wherein the slicing the portion of the audio signal corresponding to the sample into the particular number of sub-samples comprises obtaining

a linear interpolation on the sample and the next sample, wherein the particular number of sub-samples are evenly distributed on the linear interpolation.

6. The method of claim 1, wherein the first number is equal to 10.

7. The method of claim 6, wherein the particular number is equal to 1024.

8. An apparatus for audio signal processing, the apparatus comprising:
a memory; and
at least one processor coupled to the memory and configured to:
receive a data block of an audio signal comprising a first number of samples;
for each sample of the first number of samples, slice a portion of the audio signal corresponding to the sample into a particular number of sub-samples;
obtain a total number of sub-samples within the data block based on a multiplication of the first number with the particular number; and
resample the data block of the audio signal into a second number of samples that is equal to the first number plus one, when an underrun condition occurs in an electroacoustic device, or to the first number minus one, when an overrun condition occurs in the electroacoustic device;
distribute the second number of samples evenly within the total number of sub-samples obtained such that a duration corresponding to the second number of samples is equal to a duration corresponding to the first number of samples, the first number of samples and the second number of samples having been derived from the same data block; and
play back the resampled data block of the audio signal via the electroacoustic device.

9. The apparatus of claim 8, wherein the particular number is a power of 2.

10. The apparatus of claim 9, wherein arithmetic shift operations of an exponent number of bits of the data block are used instead of division operations in the resampling of the data block.

11. The apparatus of claim 8, wherein the portion of the audio signal corresponding to the sample is a portion of the audio signal between the sample and a next sample immediately subsequent to the sample.

12. The apparatus of claim 11, wherein, to slice the portion of the audio signal corresponding to the sample into the particular number of sub-samples, the at least one processor is configured to obtain a linear interpolation on the sample and the next sample, wherein the particular number of sub-samples are evenly distributed on the linear interpolation.

13. A non-transitory computer-readable medium storing computer executable code, comprising instructions for:

receiving a data block of an audio signal comprising a first number of samples; for each sample of the first number of samples, slicing a portion of the audio signal corresponding to the sample into a particular number of sub-samples;

obtaining a total number of sub-samples within the data block based on a multiplication of the first number with the particular number; and

resampling the data block of the audio signal into a second number of samples that is equal to the first number plus one, when an underrun condition occurs in an electroacoustic device, or to the first number minus one, when an overrun condition occurs in the electroacoustic device;

distributing the second number of samples evenly within the total number of sub-samples obtained such that a duration corresponding to the second number of samples is equal to a duration corresponding to the first number of samples, the first number of samples and the second number of samples having been derived from the same data block; and

playing back the resampled data block of the audio signal via the electroacoustic device.

14. The non-transitory computer-readable medium of claim 13, wherein the particular number is a power of 2.

15. The non-transitory computer-readable medium of claim 13, wherein the portion of the audio signal corresponding to the sample is a portion of the audio signal between the

sample and a next sample immediately subsequent to the sample, wherein the slicing the portion of the audio signal corresponding to the sample into the particular number of sub-samples comprises obtaining a linear interpolation on the sample and the next sample, wherein the particular number of sub-samples are evenly distributed on the linear interpolation.