

THE DEVELOPMENT OF AUDIO DATA COMPRESSION METHODS AND FACILITIES WITH THE USE OF FRACTAL APPROACH

V. Slesarev, O. Martynuk
(Ukraine, Dnipro, National Mining University)

The development of modern telecommunication networks is characterized by an increase in the share of multimedia traffic. An important component of multimedia traffic is audio information, especially voice information. This problem appeared during the working process on workstations and personal computers. The problem is associated with a large volume for their transfer and storage. There are many different algorithms for audio data archiving, but they either provide insufficient compression ratios, or lead to a significant loss of data, what is connected with deterioration of consumer information quality. According to this, there is a relevant task to develop new approaches to audio data compression.

The algorithm of conversion efficiency into a file packing is reduced to all rank sequences enumeration and selection for each corresponding domain sequence. The scheme of this algorithm is given below.

```

Step 1. The selection of the rank sequence with the length  $p$ .
Step 2. Splitting the signal  $S$  into rank and domain sequences
 $R = \text{breakToRanges}(S)$ ;
 $D = \text{breakToDomains}(S)$ ;
Step 3.
for ( $i = 1$ ;  $i \leq \text{num\_ranges}$ ;  $i++$ )
{
    for ( $j = 1$ ;  $j \leq \text{num\_domain}$ ;  $j++$ )
    {
        compute  $s$ ,  $o$ ;
        if ( $\sigma_{R_i} < \text{min\_error}$ )
        {
             $\text{min\_error} = \sigma_{R_i}$ ;
             $\text{best\_domain} = j$ ;
             $\text{best\_s} = s$ ;
             $\text{best\_o} = o$ ;
        }
    }
}

```

```

    }
  }
  save_coefficient_for_range(best_domain, best_s, best_o);
}

```

Decompression of the fractal compression algorithm is extremely simple. It is necessary to carry out several iterations of affine transformations, the coefficients of which were obtained at the stage of compression.

Any signal (e.g. zero) can be taken as an initial signal, because the corresponding mathematical apparatus guarantees the convergence of the sequence of signals obtained during IFS iterations to a stationary signal (close to the original one). Generally, 16 iterations are sufficient for this.

The coefficients of all sequences should be read from the file.

It is necessary to create a zero signal S0 of a right size.

until (the signal will not be fixed)

```

{
  R = breakToRanges(S0);
  D = breakToDomains(S0);
  for (i = 1; i ≤ num_ranges; i++)
  {
    for (j = 1; j ≤ size_range; j++)
    {
       $R_j^i = s^i \cdot D_j^{best\_domain\_i} + o^i$ ;
    }
  }
  S0 = joinRanges (R);
}

```

As a result, in this study:

1. It is demonstrated, that the use of fractal compression method is a possible and perspective way of audio data compression.
2. The mathematical substantiation of audio data fractal compression is developed. The relations for the determination of iterated functions system conversion rates are received.
3. The algorithm for fractal compression of audio data and its software implementation are developed.
4. The dependence of the efficiency of the proposed algorithm (its speed, the provided compression ratio and signal quality) on the algorithm parameters and compressible digital sound is examined.

At this stage of the algorithm for fractal compression of audio signals realization, the most effective (by quality-compression criteria) results were obtained on signals with a small difference in values between two consecutive counts. By the degree of compression, it is still far behind the existing compression algorithms.

The following steps of the existing fractal compression algorithm implementation optimization should be marked:

- the use of the Fisher algorithm, genetic algorithm, etc. instead of the search of all domain sequences;
- the saving to only shifts in value, representing a reduced copy of the encoded signal (analogy with the approximating coefficients of wavelet transformations), with their further interpolation to the original waveform restoring;
- multilevel signal decomposition - if the encoding accuracy is insufficient, the processed fragment is divided into 2 parts, each of which is processed in the same way as all the others.

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