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INVESTIGATING POLYNOMIAL APPROXIMATIONS FOR THE SPECTRA OF THE PIPE ORGAN SOUND

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An attempt was made to find some features describing the shape of the spectra of organ pipe sounds in terms of their subjective discernibility. A multi-channel recording system was employed in order to represent reliable pipe sounds in a realistic acoustic interior. A precise method for the determination of the spectral representation of pipe sounds was introduced. The polynomial approximation of the spectral envelope was found to be an effective tool, allowing the study of differences between sounds produced by organ pipes of various types belonging to some selected instruments. The paired comparison subjective testing procedure was applied in order to assess the similarities between sounds synthesized using polynomial smoothed spectra and the original organ sound patterns. The statistical processing of test results revealed that a direct relation exists between the type of organ pipe and the minimum order of the approximating polynomial that can be used to represent the pipe sound spectrum, as determined by the positive opinions of the experts. The applied pipe organ sound recording and processing methods, subjective testing procedures and experiment results are discussed in the paper.

1. Introduction

Numerous scientists have been researching the acoustic features of organ instruments for many years. The historical instruments, particularly, attract a great deal of attention. By following the publications on the subject over the last decade, one can observe an increasing number of researchers dealing with the subject. A partial reason for this is probably due to the development of measurement technique. On the other hand, the methodology of subjective testing is also observed to have been developing. Without it, any achievement in the field of acoustic instrumentation would be difficult to verify. Hence, the approach to the subject is twofold: objective analysis and auditory assessment. Much research was and still is striving to achieve homogeneity between the two concepts; that is, to come up with such rules for the interpretation of objective analysis results so that they would as much as possible be in accord with the subjective assessment. So far, subjective assessment has been the supreme authority in terms of sound quality evaluation.

Following are listed some facts that might help to give an idea of the scale of problems with regard to organs as related to the above aspects:

— Organs were constructed by organ builders in various epochs, usually in accordance with the contemporary concept determining the style of play on the instrument. However, what is essential here is not just the date of creating the instrument, but also the dates of its renovation. To be considered is the fact that a great many historical organs have been damaged in the course of history, and their later renovation was not always done in accordance with the construction rules applied by the builder of the original version of the instrument.

— Firms, which constructed organs, used various techniques and different technical solutions even though the epoch was the same. Differences include the technologies of pipe manufacture, the rules of scaling and voicing of the pipes, the selection of organ stops, the procedures of interior acoustic adaptation, etc.

— What is also extremely significant is how the room affects the acoustic image of the instrument located in it. An organ is usually fitted in the interior of a church whose characteristic feature is a long reverberation time. The acoustics of the interior affects not only the transient phases of sound, easily observed upon auditory assessment, but it also affects the steady-state phase. It is not as easy with organs as it is with other instruments to record sound for analytical purposes. It has to be taken into account that the sound is recorded in highly reverberant conditions and that it is not always in a uniformly distributed acoustic field. The outcome is a strong dependence of the measurement result on the location of the microphone recording the sound.

There are a great number of organ instruments in existence, a large percentage of which are of high historical value. Currently, most of these instruments have no acoustic documentation containing objective analyses which would allow the assessment of their sound features. This situation also concerns many instruments in Poland, where the authors performed the herein discussed investigations. The results of the research which aimed to identify a solution to the problem of objective classification and assessment of organ stops are presented in this paper.

2. Recording the organ sound

This paper focuses on the analysis of the sounds of organ pipes on the basis of their steady-states. Recording in real interiors is complicated due to a large number of acoustic wave reflections, with the number getting larger as the reverberation time gets longer for the frequency under examination. The method for the recording of consecutive organ pipes of the studied stop must, under these conditions, take into account the element of coincidence resulting from the changing of microphone position when recording individual organ stops.

The interiors of the churches where the investigated organ instruments are located have one more distinctive feature which introduces an additional obstacle to sound recording and analysis, especially when recording quiet sounds and ones containing harmonics of a low level. What is referred to here is background noise coming from external sources and from the blower that each of the instruments is equipped with. This intruding noise affects the level of the acoustic noise floor which must be considered upon interpreting the results of analysis [29]. Therefore, the value of the signal-to-noise ratio should be taken into account. Consequently, the results of spectral analysis had to be limited to such components of sound that exceeded the level of the noise floor, at least in the adjacent band range. The problem is to interpret the results of spectral analysis for harmonics of low amplitude, close to the value of the noise floor level. The spectra obtained from analysis show a loss of energy in the higher harmonics. Due to this, there is a limitation in terms of the number of spectrum components to be analyzed.

As a consequence of the above considerations, a system was adopted which was based on multi-channel organ sound recording. To reduce the effect of microphone location on the results of the analysis, it was decided to use several different recordings with microphones placed in various points. Increasing the number of recording points brings about much better accuracy in the measurements, and within limits it leads to results which exactly reflect the emitted sound. To obtain the sound spectrum that a pipe actually generates, the energy for each component would need to be summed up from the entire room. Since such an operation is clearly unfeasible and impractical, a four-channel digital recording was employed which is a compromise between the accuracy and the enormous amount of work needed for this analysis.

Recordings were made in the following Gdańsk churches: St. Mary's Basilica; St. Nicholas' Basilica; Holy Trinity Church; The Church of Jesus' Holiest Heart; and The Church of St. Peter and St. Paul. The majority of the recordings were made for the full (chromatic) musical scale, in ascending order, for the following organ stops: Principal 8′, Principal 4′, Octave 8′, Octave 4′, Subbas 16′, Flute 4′, Viola da Gamba 8′, Quintadena 8′, Trumpet 8′, Posaune 16′.

With the use of software written specially for multi-channel digital recording and sound editing on a UNIX workstation, sound files were created and then edited using selected fragments from multi-channel recordings of organ pipes belonging to the same stop. To do this, a UNIX computer was equipped with a 4-channel A/D conversion card which allowed 16-bit recording with a sampling frequency of 22.05 kHz.

3. Representation of the organ sound

A computer program written at the Sound Engineering Department of the Gdańsk University of Technology performed analyses described in this section. The first task of this program was to read sound files made after editing. Next, the program performed calculations and analysis of the steady-state spectrum on the basis of the discrete Fourier transform (DFT). When distortions caused by the background noise coming from external sources and from the blower are present in the registered signal, increasing the time window so that it encompasses several periods of the signal can provide a correction of the results of the analysis [17]. On the other hand, the DFT analysis loses accuracy when performed over several periods. To make possible the precise tuning of the time window to the period of the signal or its multiples, a variable length time window procedure was used. Data on the particular harmonics and on distortions are extracted using the above calculations. The best estimation of the signal will take place when the exact value of the fundamental frequency is known. However, in the case under discussion only an approximate value is known. This is because we must consider the possibility of organ sound mistuning with comparison to musical pitch based on the equal temperament scale. For this reason, a procedure to determine the fundamental frequency should be applied. The fundamental frequency may be obtained by changing the length of the time window in a way that leads to better tuning to the sound. After such a procedure, the spectrum and the signal-to-noise (S/N [dB]) ratio are computed according to the formula:

$$\frac{\mathrm{S}}{\mathrm{N}} = 10 \cdot \log \frac{E_S}{E_N}, \tag{3.1}$$

where:

$$E_S = \sum_{i=k}^{k \cdot n} E_i \tag{3.2}$$

represents the total energy of the harmonic spectral components, and

$$E_N = \sum_{i=1}^{k-1} \sum_{j=0}^{n} E_{i+j \cdot k} \tag{3.3}$$

represents the total energy of parasite spectral components, E_i – energy of the k-th spectrum harmonic, $E_{i+j \cdot k}$ – energy of the parasite spectral component, k – multiple of signal periods within a chosen window length, n – number of the highest spectral component.

The value of the S/N ratio is then compared to the previously computed one. The result leads to acquiring the highest possible value of the useful signal-to-noise ratio in consecutive steps until reaching the required accuracy. The main tasks performed by the developed algorithm are presented in Fig. 1. Examples of this type of analysis are presented in Fig. 2. The numbered lines in Fig. 2a and b are interpreted by the program as components of the useful signal, the others as noise. This procedure is completed for all sounds of the analyzed organ stop.

Based on the obtained results of the analyses, there are some observations useful for designing further stages of the research:

1. Spectra of the particular pipes comprising one organ stop are not identical, and they show certain discrepancies in terms of the frequency and amplitude of consecutive harmonics [13];

2. A gradual, systematic change of the spectrum is visible as one moves to extreme ends of the musical scale. This tendency is best visible for flue pipes, and is in this case the effect of scaling [13].

A statistical verification of the results was applied to the obtained data. For this purpose, a computer program was developed which performs such computations as:

• average spectrum for each organ stop, where the range of averaging may pertain to a selected part of the musical scale. This was used, for example, in researching the effect of scaling on the sound timbre. The amplitudes of components are also subjected

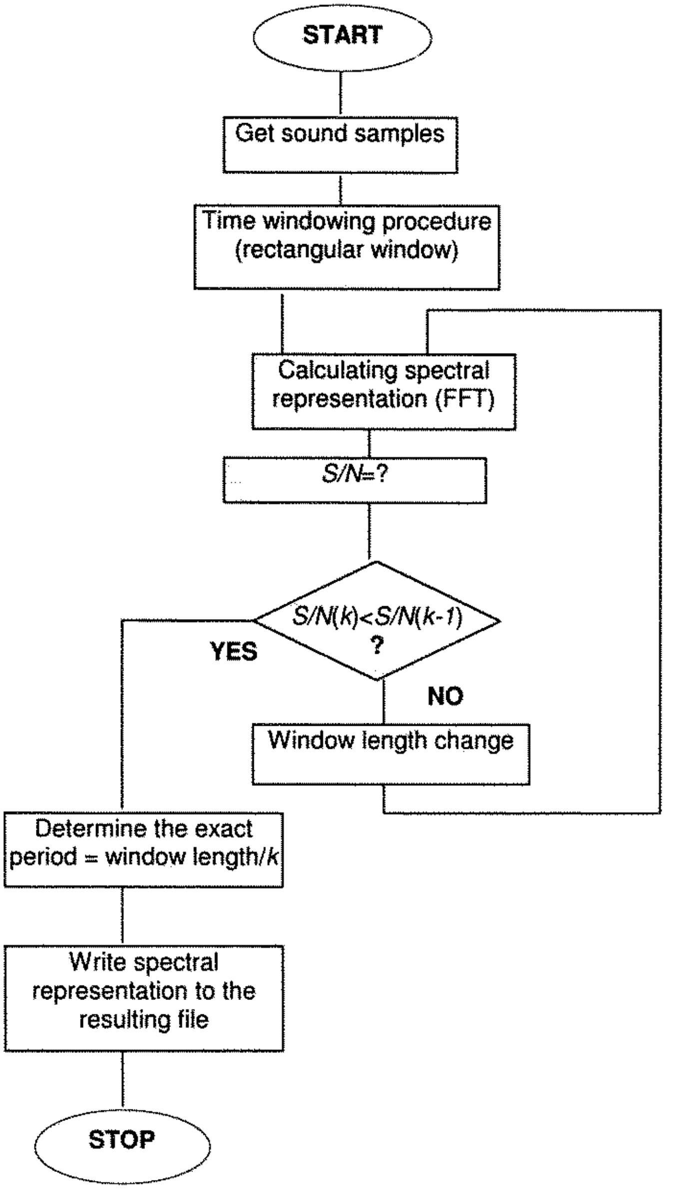


Fig. 1. Algorithm of spectrum determination using the window adjustment procedure.

to averaging computations. However, prior to averaging computations, the spectra are normalized in terms of their energy. The result is that only the relative amplitude levels of the components are considered;

• the variance computed for each harmonic is examined in order to study the dispersion in the shape of the spectrum within the given organ stop;

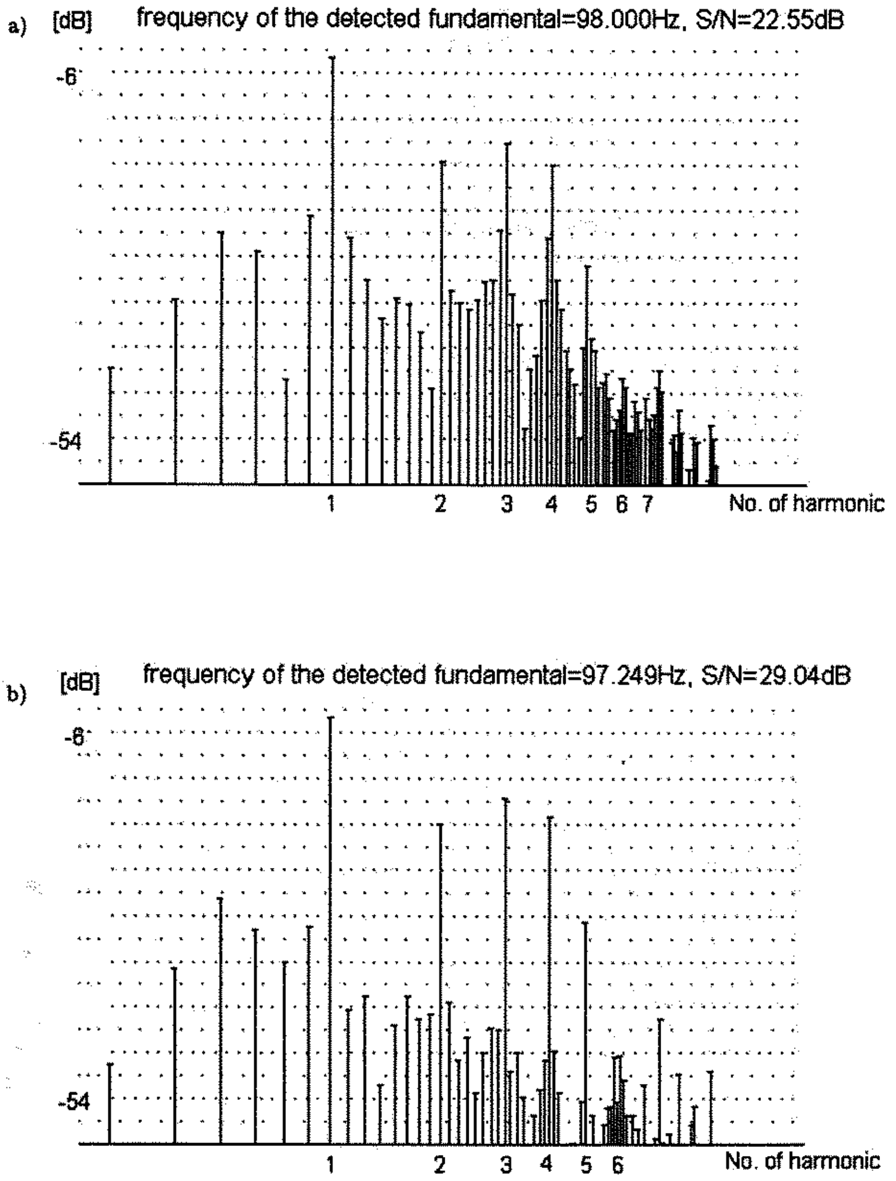


Fig. 2. Spectral analysis of the organ sounds: a) result of spectral analysis of the organ pipe sound, b) result of spectral analysis of the organ pipe sound when the tuned window length procedure was used in order to maximize the S/N ratio.

• the calculation of confidence intervals for each harmonic in order to determine estimation boundaries of the average spectrum, using the formula:

$$\frac{c_n}{2} = \frac{t^{\alpha}_{p-1} \cdot S_n}{\sqrt{p-1}}, \tag{3.4}$$

where: c_n – width of the confidence interval for the amplitude of n-th component of the average spectrum, t^{α}_{p-1} – boundary value of t Student statistics for $p-1$ degree of freedom and of significance level α, S_n – standard deviation for the amplitude of the n-th component computed from measurements according to formula (3.5):

$$S_n = \sqrt{\frac{1}{p} \cdot \sum_{i=1}^{p} \left(A_n(i) - \overline{A}_n\right)^2}, \tag{3.5}$$

where: $\overline{A}_n$ – n-th amplitude value of the n-th component of the average spectrum, n – number of the spectrum component, p – total number of sound patterns, $A_n(i)$ – n-th amplitude value of the harmonic of the i-th sound pattern spectrum after energy normalization.

The computed confidence interval for the n-th component can be written down in the form:

$$\left\langle \overline{A}_n - \frac{t^{\alpha}_{p-1} \cdot S_n}{\sqrt{p-1}}, \ \overline{A}_n + \frac{t^{\alpha}_{p-1} \cdot S_n}{\sqrt{p-1}} \right\rangle. \tag{3.6}$$

The experimentally chosen value of α for the confidence intervals was equal to 0.05. Using a graphical presentation of the computed confidence intervals for each harmonic separately, a specific confidence "sleeve" is created for the entire average spectrum.

Based on the results obtained using the above specified methods, it is possible to show that it is justified to apply the multi-channel methodology of sound analysis in an acoustic interior. To do this, it is sufficient to compare the dispersion for the particular harmonics or the widths of respective confidence intervals for the average spectrum computed from a single-channel recording with corresponding values originating from a multi-channel recording. This will be shown in next section.

4. Features of organ sound

Upon reviewing the acoustical literature, it should be noted that apart from such basic terms as loudness or pitch perception, there are several or even a dozen subjective terms for sound timbre, such as brightness, depth, sharpness, warmth, etc., which do not have such simple objective equivalents as these two basic attributes [4, 26, 32]. Some of them, however, are classified as objective through the fact that a majority of researchers agreed on a certain mathematical formula to describe them, e.g. brightness [2, 22]. Other definitions of sound parameters arose on the basis of proposals made by acousticians; they were formulated on the basis of features of musical instrument sounds [1, 3, 5, 10, 18, 32, 34].

In the case of organ sounds, the applied analytical approach can be divided into two types:

• investigating whole instruments to find common features for various schools to which the builders belonged [23, 24, 30];

• investigating particular types of pipes to identify features which are distinctive for organ stops [29, 31].

Earlier research results were the outcome of applying an analog technique where the most frequently used analyzers ensured an octave or a one-third-octave resolution in the field of the spectrum [6]. This helped to define and implement several parametric representations of sound features. In these analyses, a twofold approach can be observed: long-term averaging of sounds of the entire instrument scale or of a selected stop [11, 12, 30, 33]; or an elaborate recording of sounds from single pipes [4, 8, 27, 29]. Results of the octave analysis helped, for example, to isolate certain differences between organ instruments originating from manufacturers who belonged to various schools. The basis for this was a five-element vector of parameters consisting directly of measurement values in five octaves of the acoustic band [23, 24]. Later research showed that based on these results it is possible to reduce the length of the parameter vector while at the same time maintaining classification values. This procedure was based on the matrix of a so-called cosine transformation and eventually helped obtain a three-element vector [30]. Application of the one-third-octave analysis was used to perform feature extraction of sounds from single organ pipes by means of a division of the spectrum into groups of harmonics. This methodology led to presentation of a subjective assessment of sound timbre in terms of polar coordinates [29].

Research on the transient state produced a whole series of parameters such as rising time, releasing, overblowing, etc. [14, 15, 18, 19, 35]. Apart from these, the method of describing sound timbre evolution using a respective grouping of harmonics, called the tristimulus method, appeared [31]. The research on the transient states of musical sounds [8, 9], however, especially on the attack transients, is more closely related to articulation effects in organ playing [14, 15, 16, 18, 20, 28]. Therefore it is not the main subject of this paper. Nonetheless, the authors have participated in these types of research experiments [18, 19, 20].

The idea of describing sound timbre through a set of three parameters has recently inspired research related to sounds coming from recordings of a whole musical group [25]. In this work, however, the main considerations pertain to single organ pipes. In the case of such sounds, there is a well-known method of describing this feature, called spectrum irregularity (French: *irregularité du spectre*) [22].

As will be shown in the next section, builders of organ instruments were most probably guided by the acoustic effect related to the sound spectrum when seen as a whole. For example, the sound timbre can be described through defining the size of decay of the higher harmonics in relation to the lower ones. In this case, a parameter may be proposed: the directional coefficient of a line adjusted for a set of points matching the maxima of the harmonics, expressed in $\mathrm{dB}/\log f$. Application of this linear approximation of the spectrum may be the point of departure to a better description of sound timbres of organ pipes. Feature extraction based on groups of harmonics is in a way a reference to such an approach [31]. However, using this extraction as a basis, it is difficult to perform additive synthesis which for the steady-state requires not only the knowledge of what the fundamental frequency is, but also the amplitudes of the particular components, and in some cases the knowledge of what the phase variations are. Therefore, it seems that it will be more beneficial to identify a more accurate description of the envelope of the sound spectrum.

The authors' proposal of such a description is based on an approximation of the spectral envelope shape using polynomials [13, 17]. This approach seems to be correct, especially in applications involving richer spectra such as the sound spectrum of reed pipes. The applied approximation is based on minimizing the mean-square error in the range of the analyzed spectrum, and is computed by using the following proposed relation:

$$E = \sqrt{\sum_{i=1}^{m}(20 \cdot \log_{10} A(i) - W_r(\log_2 i))^2}\,, \tag{4.1}$$

while:

$$E_r(\log_2 i) = \sum_{j=0}^{r} a_j \cdot (\log_2 i)^j, \tag{4.2}$$

where: E – mean-square error, i – number of the harmonic, $i = 1, 2, ...m$, m – number of the highest harmonic, $A(i)$ – value of the amplitude of i-th component, $W_r(\log_2 i)$ – value of the polynomial for i-th component, a_j – j-th term of the polynomial, j – number of the consecutive term of the polynomial, r – order of the polynomial.

Computations which minimize the error are performed by consecutive substitution of $r = 1, 2, \ldots$etc. up to the assumed maximum value, successively obtaining the coefficients $a_1, a_2, a_3, \ldots$. Based on formula (4.1), the approximation is performed on the spectrum presented in the log-log scale, which causes the consecutively computed coefficients a_j, where $j = 1, 2, 3, \ldots$ to be expressed respectively in dB/octave, dB/octave^2, dB/octave^3, etc. These coefficients have a clear physical interpretation, e.g. the first defines the decay of higher harmonics in the spectrum, whereas the second indicates a gain or a loss of the middle part of the spectrum in relation to its lower or higher parts.

Moreover, by raising the approximation order, more coefficients are obtained which describe more precisely the spectrum of the real sound. Such examples of an ascending order polynomial approximation of the steady-state spectrum of an organ pipe sound pattern will be shown in the next section.

5. Statistical analysis of the recorded material

To determine whether the computed parameters – coefficients of polynomials approximating the spectrum – can be treated as distinctive features, a statistical tool was used in the study, in the form of Behrens-Fisher's statistics [7, 13]. The basic assumption is that of mean equality in two normally distributed populations. The following values are computed from the formulae:

- mean parameter values:

$$\overline{X} = \frac{1}{n} \cdot \sum_{i=1}^{n} X_i\,, \qquad \overline{Y} = \frac{1}{m} \cdot \sum_{i=1}^{m} Y_i\,, \tag{5.1}$$

- and variance estimators of respective random variables:

$$S_1^2 = \frac{1}{n-1} \cdot \sum_{i=1}^{n} \left(X_i - \overline{X}\right)^2, \qquad S_2^2 = \frac{1}{m-1} \cdot \sum_{i=1}^{m} \left(Y_i - \overline{Y}\right)^2, \tag{5.2}$$

where: n, m – cardinality of populations X and Y accordingly, $S_1, S_2 \neq 0$.

On the basis of the values obtained from Eq. (5.1) and (5.2), the statistics are computed from the following relation:

$$V = \frac{\overline{X} - \overline{Y}}{\sqrt{S_1^2/n + S_2^2/m}}, \tag{5.3}$$

which can then be compared to a respective boundary value from statistical tables under the assumed significance level. This requires additional computation of the statistical parameter c from the formula:

$$c = \frac{S_1^2/n}{S_1^2/n + S_2^2/m}, \tag{5.4}$$

which reflects the distribution of statistics V and the determination of the boundary value.

Statistics V, computed for each parameter separately, can also be treated as a measure of the distance between the compared classes in the analyzed space of parameters. It is this interpretation that was also applied in the described experiments. For the assumed and fixed n and m, it is possible to compare the values computed for various parameters related to the investigated populations.

The above described test using Behrens-Fisher's statistics was applied to study the multi-channel methodology, which can also be examined using the confidence interval method described in Sec. 3. The latter method is illustrated in Fig. 3. The figures present the average spectra of selected types of pipes from the organ at St. Mary's Basilica in Gdańsk. For each spectral component, a confidence interval was found at a level of 0.95. The range of sounds was reduced to one octave, in this case the 12 sounds of the chromatic scale from C2 to B2. The spectra in Fig. 3 are presented in the log-log scale. The x-axis is related to normalized frequency in terms of the frequency of the fundamental component. The y-axis represents the normalized energy of the particular harmonics expressed in [dB]. Comparing the width of confidence intervals computed for average spectra of both single and four-channel recording proves the advantages of applying multi-channel recording. In the latter case the confidence interval width is much narrower than in the case of a single channel recording, what signifies a smaller dispersion of computed spectral components (see Fig. 3).

This observation is further confirmed by the results of the statistical analysis (Table 1) and analysis, which is based on the Behrens-Fisher's statistics (Fig. 4). The approximation parameters which were used as initial data in the test were computed on the basis of the same results. In the latter case, a limitation of four stops was introduced. The obtained results show that four-channel recording usually leads to a smaller dispersion and consequently to a better separation between the parameter values in comparison to single-channel recording.

To show the differences in the spectra of sounds from various parts of the musical scale, average spectra were computed and feature extraction was performed for these sounds. This has been illustrated in Fig. 5 in graphical form as a diagram of an approximating polynomial in points matching spectral lines. Examples of the results of the

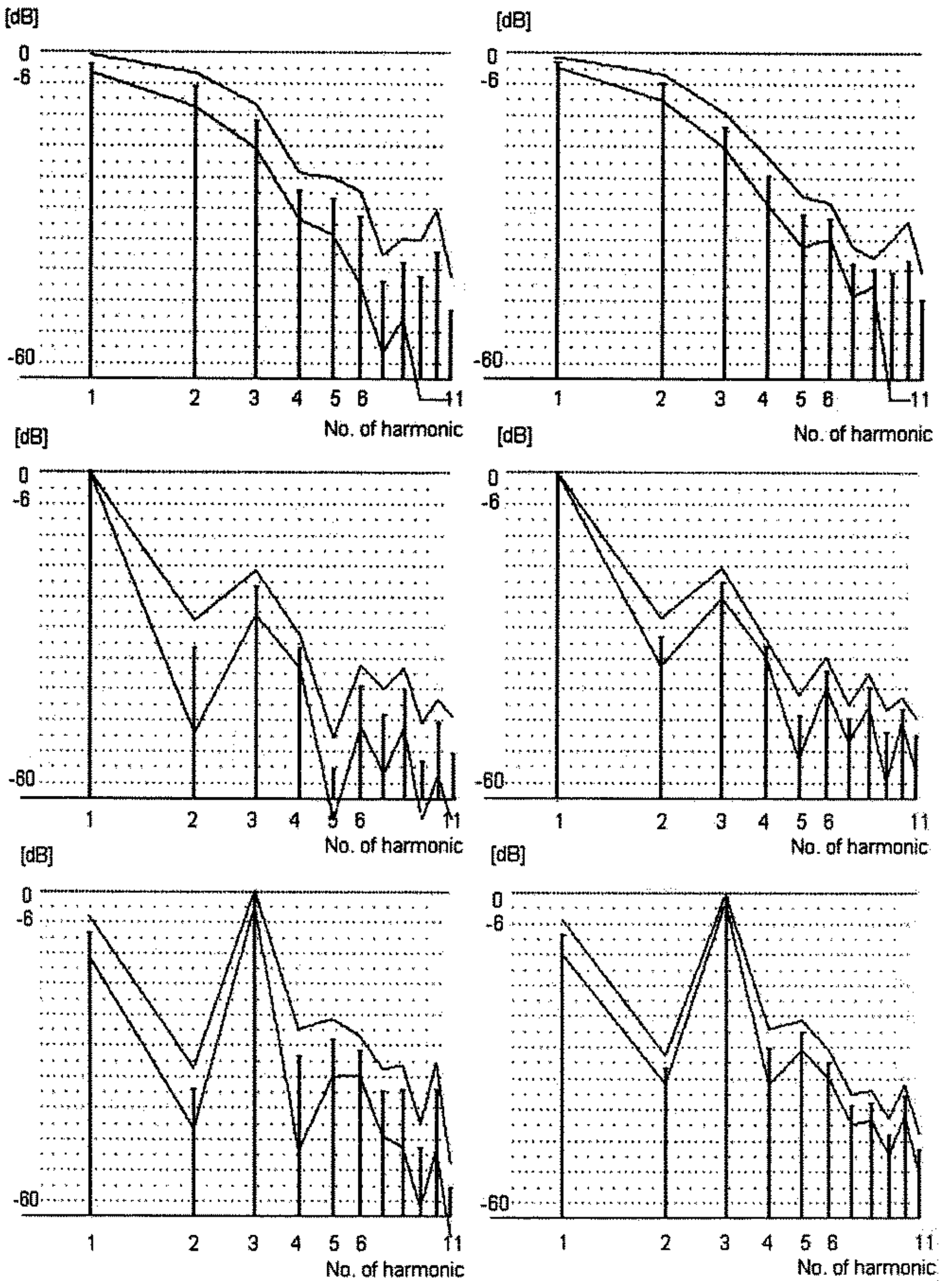


Fig. 3. Comparison between average spectra of sounds from the chromatic scale of the octave from C2 to B2 of three selected flue pipe organ stops at St. Mary's Basilica in Gdańsk. Going from the top, in successive order: Principal 8′; Subbas 16′; and Quintadena 8′. On the left, the result based on a single channel recording; on the right, based on a four-channel recording.

Table 1. Average values and dispersions for parameters obtained on the basis of single- and four-channel recording.

Principal 8′ – single-channel recording					
Parameter No.	1	2	3	4	5
mean value	−17.092	−2.621	2.138	−0.111	1.531
dispersion	3.454	3.458	2.558	2.139	4.322
Principal 8′ – four-channel recording					
Parameter No.	1	2	3	4	5
mean value	−15.542	−2.085	1.805	−0.377	−1.003
dispersion	2.415	2.074	1.007	1.421	1.440

analysis based on the Behrens-Fisher's statistics for the stops originating from different instruments are given in Table 2.

Table 2. Results of the comparisons of sound parameters between the same stops from various organ instruments using Behrens-Fishers' statistics.

Principal 8′: object I – object IV					
Parameter No.	1	2	3	4	5
Statistics V	1.488	−1.385	−1.501	0.655	1.479
c parameter	0.420	0.330	0.135	0.169	0.171
Principal 4′: object IV – object III					
Parameter No.	1	2	3	4	5
Statistics V	**9.590**	−0.581	0.315	−0.236	−2.810
c parameter	0.419	0.166	0.167	0.116	0.269
Posaune 16′: object I – object II					
Parameter No.	1	2	3	4	5
Statistics V	**4.671**	1.622	0.391	0.466	−2.415
c parameter	0.387	0.567	0.622	0.387	0.612
Posaune 16′: object I – object III					
Parameter No.	1	2	3	4	5
Statistics V	2.513	2.687	−0.207	**4.700**	−0.726
c parameter	0.179	0.385	0.827	0.138	0.465
Posaune 16′: object II – object III					
Parameter No.	1	2	3	4	5
Statistics V	−0.634	1.417	−0.800	**4.254**	1.514
c parameter	0.257	0.323	0.744	0.202	0.356
Subbas 16′: object II – object I					
Parameter No.	1	2	3	4	5
Statistics V	−1.702	−1.041	−1.301	1.643	1.174
c parameter	0.527	0.684	0.328	0.355	0.441

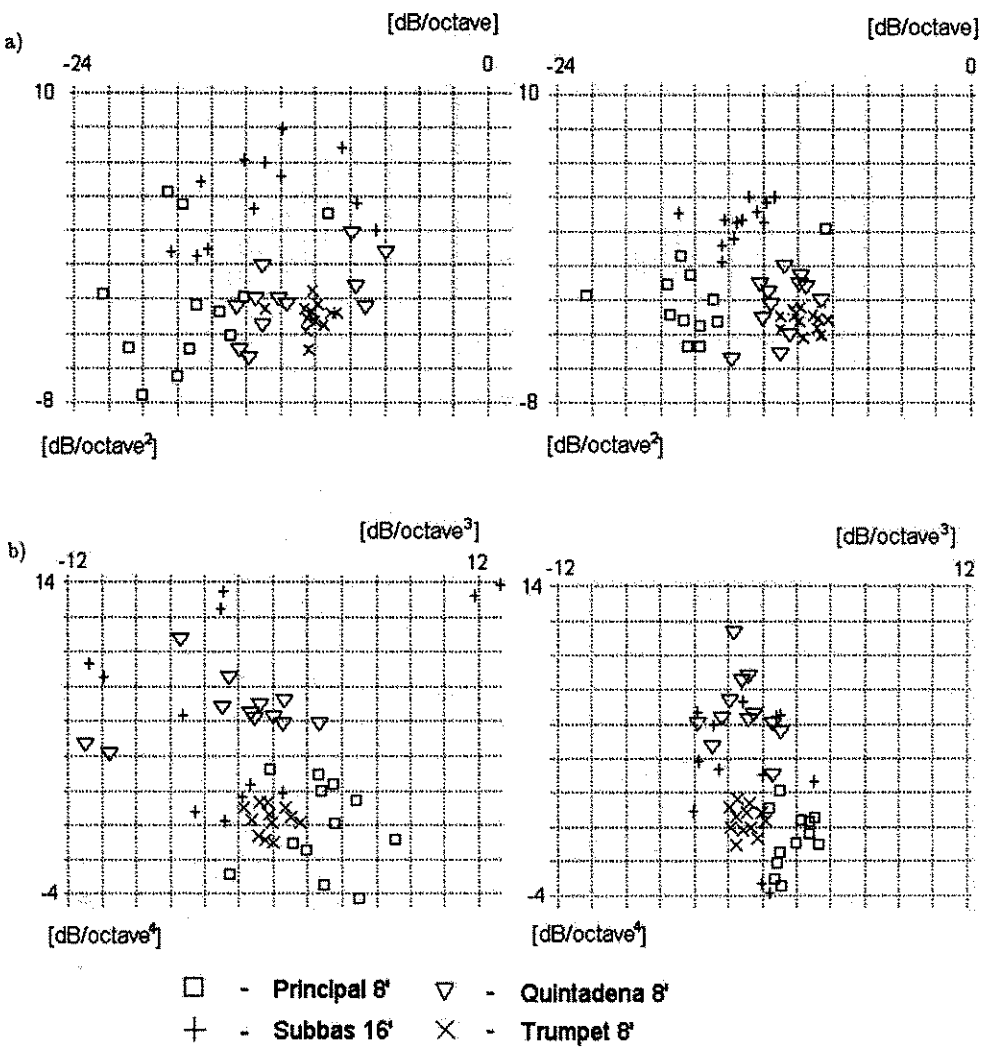


Fig. 4. Graphical presentation enabling the comparison of parameter distribution for the analyzed sounds – on the left, single-channel recording; on the right, four-channel recording: a) compared parameters are coefficients which were previously introduced: namely, linear, expressed in dB/octave; and squared, expressed in dB/octave^2, b) compared parameters are coefficients which were previously introduced: namely, 3-rd order, expressed in dB/octave^3; and 4-th order, expressed in dB/octave^4.

The chromatic scale of the octave from C3 to B3 (12 sounds) was studied for four and eight foot stops, and the chromatic scale of the octave from C2 to B2 (12 sounds) for sixteen foot stops (see Table 2). The compared stops come from organs located in Gdańsk churches, according to the following symbols: object I – St. Mary's Basilica; object II – St. Nicholas' Basilica; object III – The Church of St. Peter and St. Paul; object IV – The Church of Jesus' Holiest Heart. Values for which the value of statistics V exceeded the range from 2.845 to 3.169 were distinguished. The following assumptions

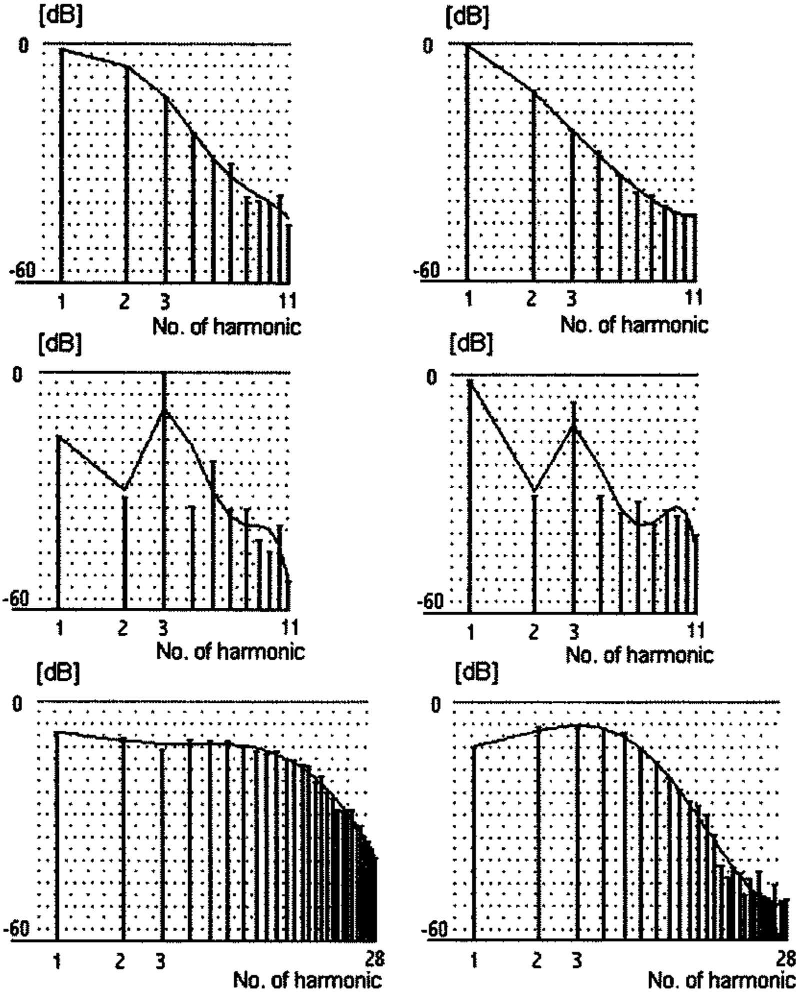


Fig. 5. Comparison between the average spectra of sounds from the chromatic scale of the octave from C2 to B2, and the octave from C5 to B5 of three selected organ stops at St. Nicholas' Basilica in Gdańsk. Going from the top, in successive order: Principal 8′; Quintadena 8′; and Trumpet 8′. On the left, the octave from C2 to B2; on the right, the octave from C5 to B5. Envelopes of the spectra are approximated with a 5-th order polynomial. Axes x and y are scaled like in Fig. 2.

were adopted: the statistical parameter c ranged from $(0, 1)$; the number of freedom degree 10; the significance level was adopted at 0.01.

The applied experiments showed that for the majority of parameters, the assumption of normal distribution of the investigated populations was correct. Some exceptions exist, however, such as: parameter 1, upon comparing Principal 4′ of objects IV with III; parameter 1, upon comparing Posaune 16′ of objects I with II; and parameter 4, upon comparing Posaune 16′ of objects I with III and II with III. This may be the result of differences due to construction details in the compared organ instruments.

6. Subjective verification

The paired comparison, non-parametric test procedure was applied [21] in order to verify whether the applied system of feature extraction provides parameters related to subjective sound perception. In the case of such a test, it was not necessary to determine particular features to be assessed by the experts (they produced overall preference scores).

For this purpose, digitally recorded real signals were used as signal patterns, and signals synthesized on the basis of additive synthesis of approximation parameters were used as test signals. The sounds used for comparison had spectra computed using approximating polynomials of orders within the range from one to five. An elementary test was to provide the real signal as the pattern and two signals synthesized on the basis of various approximation orders. All the signals had duration of 2 seconds, and the intervals between them were 0.5 seconds. The intervals between the elements of the test were 5 seconds. This was also the time designated for a subject response. The test question was, "*Which of the two synthesized signals has a timbre more similar to the model signal pattern?*"

On the basis of the experts' answers, a preferential diagram was obtained which turned out to be a rising curve for all the tests, showing a preference for the higher order approximation. Resulting from the statistical analysis of the results, the significance of this preference was also calculated. This value indirectly provides information on the approximation order that realizes a good reproduction of the investigated sound timbre. On this basis, it was possible to determine the saturation point of the preference curve.

Sounds of three various types of pipes were selected for testing: Principal; Viola da Gamba; and Trumpet. Two musical notes were used, C and B, with corresponding fundamental frequencies at about 130.8 and 246.9 Hz. The applied approximation of the order from 1 to 5 led to producing 10 pairs of sounds for one part of the test. As a result, a single testing session presented 20 pairs for comparison. The tests were carried out for two groups of experts, with a total number of 22 people. Altogether 12 test sessions were carried out.

Figures 6 and 7 illustrate graphically the exemplary spectra selected for subjective tests: real and synthetic ones.

Analysis of the results obtained as the end product of the tests was performed using a program written in the C language on a UNIX computer. Sets of answers from the experts provided direct test results that were then analyzed statistically. The schema of computations is the following:

- Summing up the number of votes given by the particular experts for each of the objects,
- Determining the stability of each experts' choices (parameter z_1),
- Determining the sum of votes given to an object by the successive experts,
- Determining the number of votes given to an object by successive experts to both parts of the test,
- Determining the statistics χ^2 by comparing the results of both parts of the test,

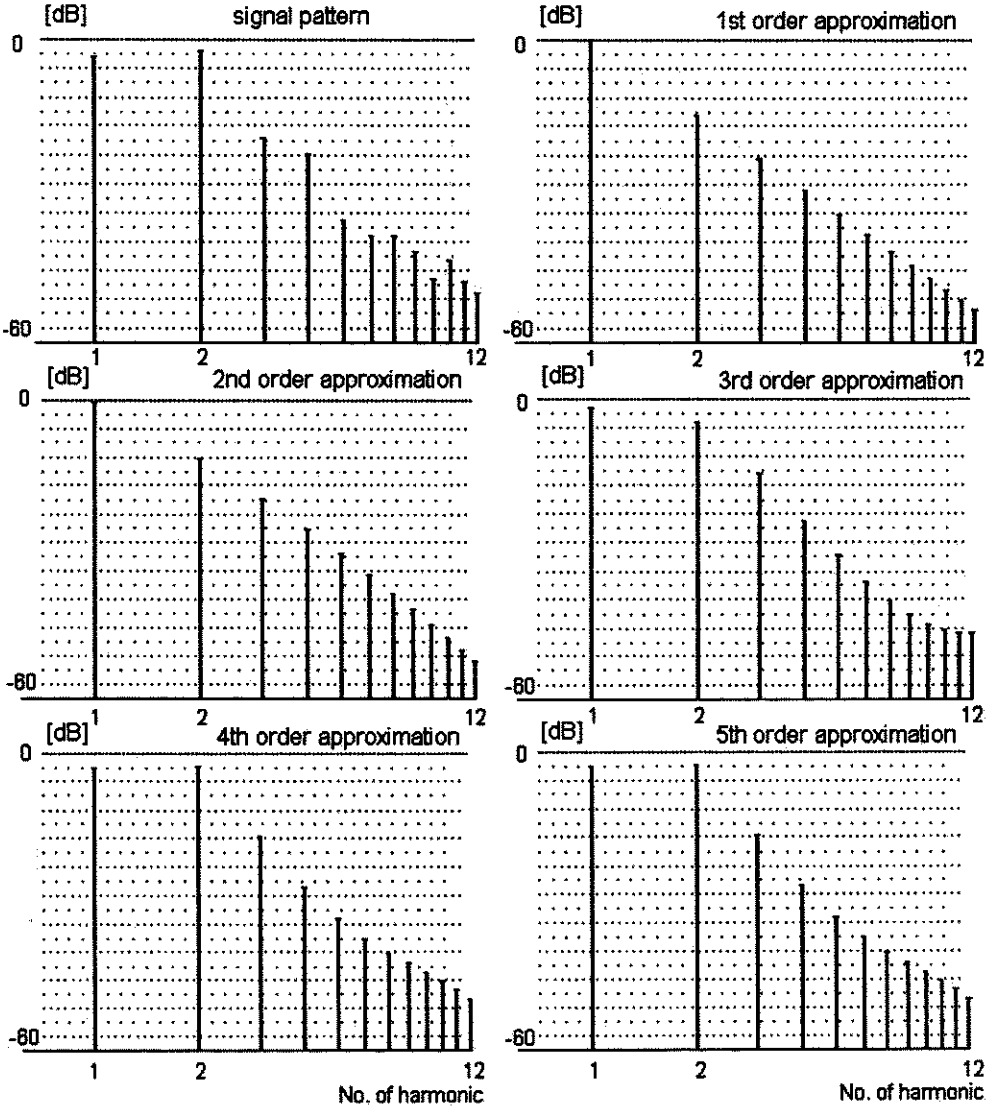


Fig. 6. Spectra of sounds used in subjective test No. 1: the pattern signal (the note C from the Principal 8′ organ stop at St. Mary's Basilica) at the left upper end; and the sounds generated on the basis of the polynomial approximation results, with rising polynomial order.

• Determining the number of experts who interpret a given pair in a different way depending on the part of the test (parameter z_2),

• Examining the significance of the differences between the objects in a pair at the assumed level of significance (parameter z_3),

Since the test was carried out for two groups of experts, the number of results was doubled and comparison between the groups becomes possible through determining the

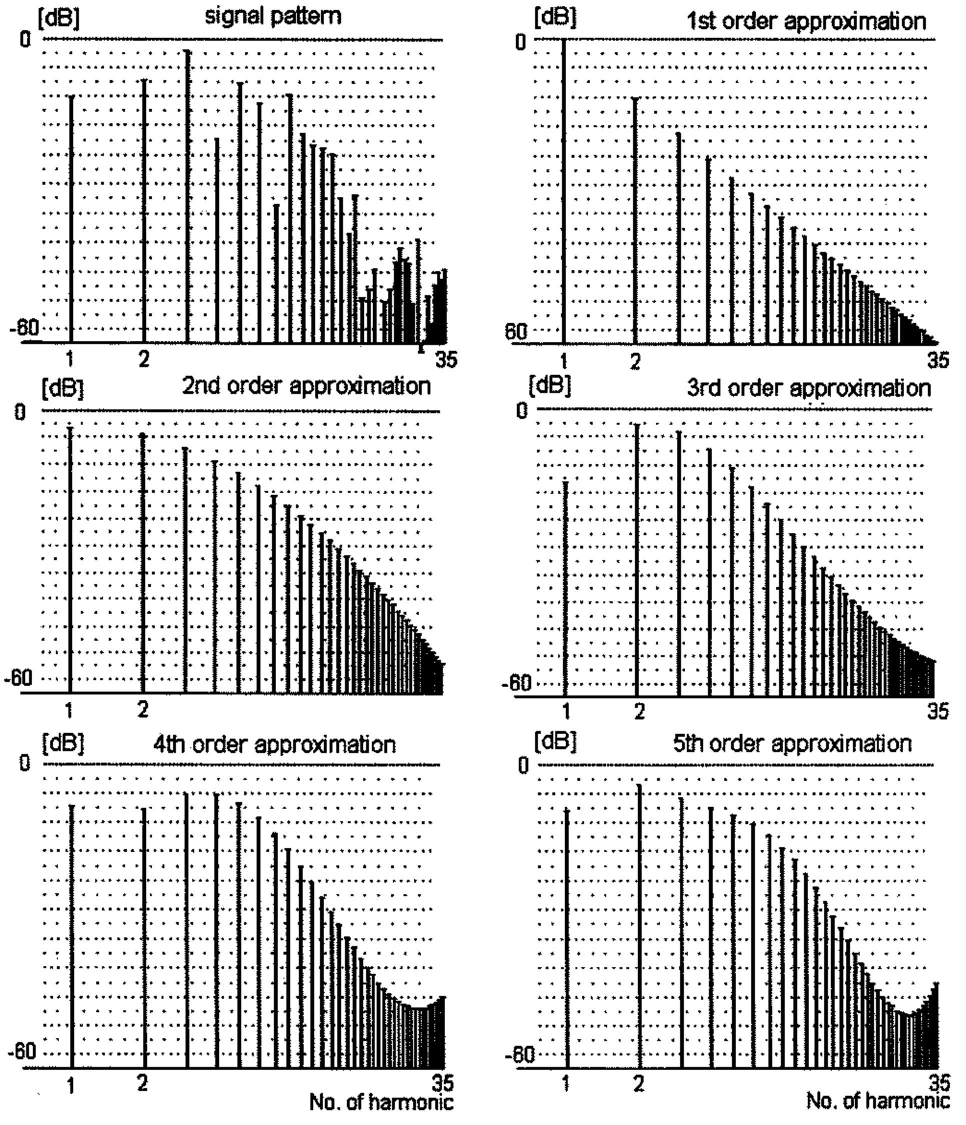


Fig. 7. Spectra of sounds used in subjective test No. 1: the pattern signal (the note B from the Trumpet 8′ organ stop at St. Mary's Basilica) at the left upper end; and the sounds generated on the basis of the polynomial approximation results, with rising polynomial order.

statistics χ^2 by comparing the results obtained from the two groups of experts with each other.

Examples of results from test 1 are given in Table 3. The upper part of the table contains the particular numbers of votes obtained by the objects. A column for parameter z_1 was distinguished – the number of errors made by each expert. It also shows the total sums of votes for the objects, and also for parts of the tests. The lower part of the

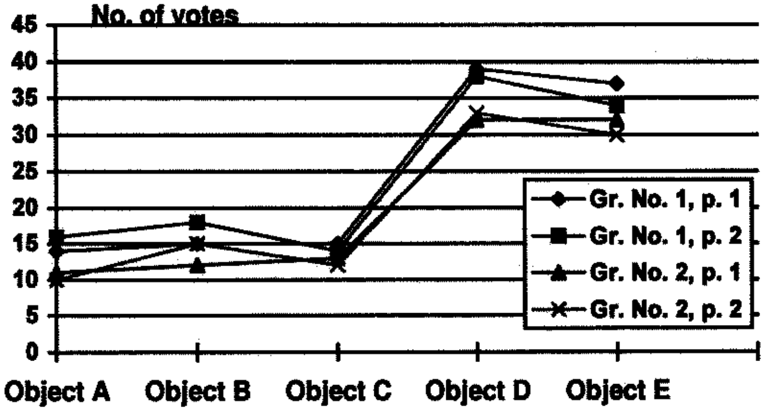
Table 3. Results of test No. 1 for group No. 1.

OBJECT	A	B	C	D	E	z_1
Expert 1	3	3	2	8	4	3
Expert 2	5	3	1	6	5	3
Expert 3	1	1	6	6	6	1
Expert 4	4	2	1	8	5	1
Expert 5	3	2	2	6	7	1
Expert 6	1	4	4	6	5	2
Expert 7	3	1	4	8	4	3
Expert 8	4	3	0	7	6	3
Expert 9	1	2	4	5	8	2
Expert 10	3	2	2	6	7	2
Expert 11	1	3	4	5	7	3
Expert 12	1	3	3	6	7	2
sum	30	29	33	77	71	240
part 1	14	15	15	39	37	120
part 2	16	14	18	38	34	120

PAIR	AB	AC	AD	AE	BC	BD	BE	CD	CE	DE
z_2	6	3	0	3	1	1	3	2	5	2
z_3	–	–	+	+	–	+	+	+	+	–

table shows the values of parameters z_2 and z_3 for all pairs of the compared objects. Parameter z_2 gives the total number of errors for a given pair, whereas for parameter z_3 the mark + was given in the case of meeting the criterion of differences significance; that is, exceeding the boundary value of 1.96 by the value of statistics. The results were also depicted in the form of examples of preference diagrams in Fig. 8. Successive objects from A to E are synthesized sounds related to the rising approximation order from 1 to 5.

To verify the assumption with regard to the effect of auditory memory on perception, a comparison of the results from both parts of the test had to be carried out. A similar comparison of the results from the two groups of experts provided the answer to the question of conformity of the observed tendencies. Both of these comparisons were performed using Pearson's test χ^2. This is one of the methods used to study the independence of features of population elements. A comparison of the results of the auditory monitoring tests between the groups of experts, which show conformity of interpretation of the tested sounds, is presented in Table 4. Additionally, a comparison between the results of both parts of the tests shows a slight effect of auditory memory on the ability to differentiate the timbre of the tested sounds (Table 5).



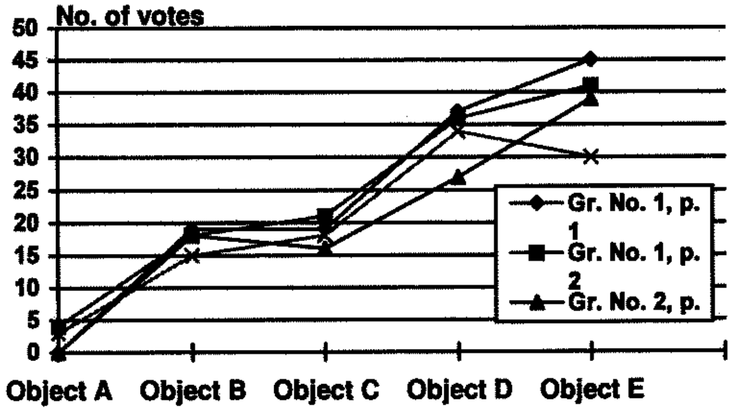


Fig. 8. Examples of preference diagrams for assessment tests: at the top, test No. 1; below, test No. 6.

Table 4. Comparison of the results of the auditory tests between groups of experts.

	Test 1	Test 2	Test 3	Test 4	Test 5	Test 6
Test 1		**12.976**	**14.052**	**26.703**	**28.063**	**15.974**
Test 2	**17.247**		8.742	4.328	**14.062**	7.217
Test 3	**19.417**	**11.660**		**21.058**	**11.613**	**11.197**
Test 4	**31.414**	2.681	**18.962**		**20.601**	**13.928**
Test 5	**39.943**	**18.233**	**13.463**	17.798		5.944
Test 6	**23.063**	9.913	**15.034**	**13.336**	**11.096**	

Table 5. Comparison between the results of both parts of the tests.

	Group 1	Group 2
Test 1	0.580	0.501
Test 2	3.114	4.466
Test 3	3.180	1.737
Test 4	8.588	5.022
Test 5	1.366	0.725
Test 6	4.327	5.368

7. Results and conclusions

The presented system of sound feature extraction based on smoothing the spectrum of the steady-state through polynomial approximation allows the comparison of sounds produced by various organ pipes. Its effectiveness was confirmed through statistical analysis of databases obtained as the result of feature extraction with this method, and also, perhaps more importantly, through the results of non-parametric subjective tests. The obtained preference curves are usually of a rising character, showing an increase in the similarity of the sound timbre together with an increase in the order of polynomial approximation. The exceptions to this tendency are local, pertain only to the final part of the diagram and are statistically insignificant. Also observed is the effect of saturation of preference curves, different for various types of pipes. This effect describes when the preference curve reaches a state, after an initial fast growth, in which a further change becomes statistically insignificant. There are various manifestations for various stops, and a statistically significant difference was found between reed and flue stops. These results are mutually confirmed by both groups of experts. The polynomials approximating spectral envelopes enable synthesis of sound that imitates the analyzed real sound. The fact that there is a saturation point of the preference curve may suggest that a significant similarity has been reached between the timbre of sounds synthesized on the basis of the polynomial spectrum representation and the timbre of the original sounds. Consequently, it is possible to reduce the approximation order to the number tested in the study, or to one that is not much greater.

The system of feature extraction developed in this study does not eliminate the possibility of extending the analytical apparatus for the purpose of sound classification. It is also possible to modify this system in order to better approximate the spectrum. Especially interesting is the concept of spectrum division into two parts, even and odd, and then separate approximations of their envelopes. There is a significant number of stopped pipes which as a rule should not generate any even harmonics. In this case, an attempt to approximate the entire spectrum will probably result in a bigger error than after its division into the even and odd parts. Such an approximation would require the application of two polynomials, but it is nonetheless probable that the final vector of parameters would include fewer elements while securing a better accuracy of approximation. Moreover, a wavelet transform can provide a platform for representing some spectral properties of organ pipe sounds more accurately than a Fourier transform. These are topics of planned future research.

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