Automated Quality Control in Sound Speaker Manufacturing using Soft Computing Techniques and Fractal Theory for Pattern Recognition

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Abstract

We describe in this paper the application of a hybrid neuro-fuzzy-fractal approach to the problem of automated quality control in sound speaker manufacturing. Traditional quality control has been done by manually checking the quality of sound after production. This manual checking of the speakers is time consuming and occasionally was the cause of error in quality evaluation. For this reason, we developed an intelligent system for automated quality control in sound speaker manufacturing. The intelligent system has a fuzzy rule base containing the knowledge of human experts in quality control. The parameters of the fuzzy system are tuned by applying the ANFIS methodology using, as training data, a real time series of measured sounds as given by good sound speakers. We also use the fractal dimension as a measure of the complexity of the sound signal. The intelligent system has been tested in a real plant with very good results.

Key words: Fuzzy Logic, Neural Networks, Neuro-Fuzzy, Manufacturing, Fractal Theory

1. Introduction

We describe in this paper the application of a neuro-fuzzy-fractal approach to the problem of quality control in the manufacturing of sound speakers in a real plant. The quality control of the speakers was done before by manually checking the quality of sound achieved after production [4]. A human expert evaluates the quality of sound of the speakers to decide if production quality was achieved. Of course, this manual checking of the speakers is time consuming and occasionally was the cause of error in quality evaluation [7]. For this reason, it was necessary to consider automating the quality control of the sound speakers. The problem of measuring the quality of the sound speakers is as follows:

1) First, we need to extract the real sound signal of the speaker during the testing period after production
2) Second, we need to compare the real sound signal to the desired sound signal of the speaker, and measure the difference in some way
3) Third, we need to decide on the quality of the speaker based on the difference found in step 2. If the difference is small enough then the speaker can be considered of good quality, if not then is bad quality.

The first part of the problem was solved by using a multimedia kit that enable us to extract the sound signal as a file, which basically contains 108000 points over a period of time of 3 seconds (this is the time required for testing). We can say that the sound signal is measured as a time series of data points [3], which has the basic characteristics of the speaker. The second part of the problem was solved by using a neuro-fuzzy approach to train a fuzzy model with the data from the good quality speakers [9]. We used the ANFIS approach [6] to obtain a Sugeno fuzzy system [13] with the time series of the ideal speakers. In the ANFIS approach a neural network [5, 10, 12] is used to adapt the parameters of the fuzzy system with real data of the problem. With this fuzzy model, the time series of other speakers can be used as checking data to evaluate the total error between the real speaker and the desired one. The third part of the problem was solved by using another set of fuzzy rules [14], which basically are fuzzy expert rules to decide on the quality of the speakers based on the total checking error obtained in the previous step. Of course, in this case we needed to define membership functions for the error and quality of the product, and the Mamdani reasoning approach was used. We also use as input variable of the fuzzy system the fractal dimension of the sound signal. The fractal dimension [8] is a measure of the geometrical complexity of an object (in this case, the time series). We tested our neuro-fuzzy-fractal approach for automated quality control during production with real sound speakers with excellent results. Of course, to measure the efficiency of our intelligent system we compared the results of the neuro-fuzzy-fractal approach to the ones by real human experts. The results clearly show that the neuro-fuzzy-fractal approach was better than the manual method because it reduced the time required for testing and also the accuracy was improved slightly. We think that our neuro-fuzzy-fractal approach for quality control can be used for similar problems, with only some minor changes to the structure of the fuzzy system.

2. Basic Concepts of Sound Speakers

In any sound system, ultimate quality depends on the speakers [4]. The best recording, encoded on the most advanced storage device and played by a top-of-the-line deck and amplifier, will sound awful if the system is hooked up to poor speakers. A system's speaker is the component that takes the electronic signal stored on things like CDs, tapes and DVD's and turns it back into actual sound that we can hear.

2.1 Sound Basics

To understand how speakers work, the first thing you need to do is understand how sound works. Inside your ear is a very thin piece of skin called the eardrum. When your eardrum vibrates, your brain interprets the vibrations as sound. Rapid changes in air pressure are the most common thing to vibrate your eardrum.

An object produces sound when it vibrates in air (sound can also travel through liquids and solids, but air is the transmission medium when we listen to speakers). When something vibrates, it moves the air particles around it. Those air particles in turn move the air particles around them, carrying the pulse of the vibration through the air as more and more particles are pushed farther from the source of the vibration.

To see how this works, let's look at a simple vibrating object -- a bell. When you ring a bell, the metal vibrates -- flexes in and out -- rapidly. When it flexes out on one side, it pushes out on the surrounding air particles on that side. These air particles then collide with the particles in front of them, which collide with the particles in front of them, and so on. When the bell flexes away, it pulls in on these surrounding air particles, creating a drop in pressure that pulls in on more surrounding air particles, which creates another drop in pressure that pulls in particles that are even farther out, and so on. This decreasing of pressure is called rarefaction.

In this way, a vibrating object sends a wave of pressure fluctuation through the atmosphere. When the fluctuation wave reaches your ear, it vibrates the eardrum back and forth. Our brain interprets this motion as sound. We hear different sounds from different vibrating objects because of variations in:

- sound wave frequency -- A higher wave frequency simply means that the air pressure fluctuates faster. We hear this as a higher pitch. When there are fewer fluctuations in a period of time, the pitch is lower.
- air pressure level -- the wave's amplitude -- determines how loud the sound is. Sound waves with greater amplitudes move our ear drums more, and we register this sensation as a higher volume.

A speaker is a device that is optimized to produce accurate fluctuations in air pressure.

A microphone works something like our ears. It has a diaphragm that is vibrated by sound waves in an area. The signal from a microphone gets encoded on a tape or CD as an electrical signal. When you play this signal back on your stereo, the amplifier sends it to the speaker, which re-interprets it into physical vibrations. In the next section, we'll see how the speaker accomplishes this task.

2.2 Making Sound

In the last section we saw that sound travels in waves of air pressure fluctuation, and that we hear sounds differently depending on the frequency and amplitude of these waves. We also learned that microphones translate sound waves into electrical signals, which can be encoded onto CDs, tapes, LPs, etc. Players convert this stored information back into an electric current for use in the stereo system.

A speaker is essentially the final translation machine -- the reverse of the microphone. It takes the electrical signal and translates it back into physical vibrations to create sound waves. When everything is working as it should, the speaker produces nearly the same vibrations that the microphone originally recorded and encoded on a tape, CD, LP, etc. Traditional speakers do this with one or more drivers. A driver produces sound waves by rapidly vibrating a flexible cone, or diaphragm. Figure 1 shows a typical speaker driver.

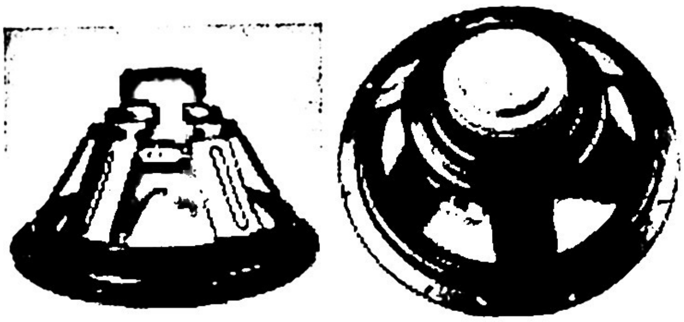


Figure 1 A typical speaker driver, with a metal basket, heavy permanent magnet and paper diaphragm

- The cone, usually made of paper, plastic or metal, is attached on the wide end to the suspension, or surround.
- This rim of flexible material allows the cone to move, and is attached to the driver's metal frame, called the basket.
- The narrow end of the cone is connected to the voice coil.
- The coil is attached to the basket by the spider, a ring of flexible material. The spider holds the coil in position, but allows it to move freely back and forth.

Some drivers have a dome instead of a cone. A dome is just a diaphragm that extends out instead of tapering in.

The voice coil is a basic electromagnet. An electromagnet is a coil of wire, usually wrapped around a piece of magnetic metal, such as iron. Running electrical current through the wire creates a magnetic field around the coil, magnetizing the metal it is wrapped around. The field acts just like the magnetic field around a permanent magnet: It has a polar orientation -- a "north" end and a "south" end -- and it is attracted to iron objects. But unlike a permanent magnet, in an electromagnet you can alter the orientation of the poles. If you reverse the flow of the current, the north and south ends of the electromagnet switch.

This is exactly what a stereo signal does -- it constantly reverses the flow of electricity. If you've ever hooked up a stereo system, then you know that there are two output wires for each speaker -- typically a black one and a red one. Figure 2 shows the wire that runs through the speaker system.

Essentially, the amplifier is constantly switching the electrical signal, fluctuating between a positive charge and a negative charge on the red wire. Since electrons always flow in the same direction between positively charged particles and negatively charged particles, the current going through the speaker moves one way and then reverses and flows the other way. This alternating current causes the polar orientation of the electromagnet to reverse itself many times a second.

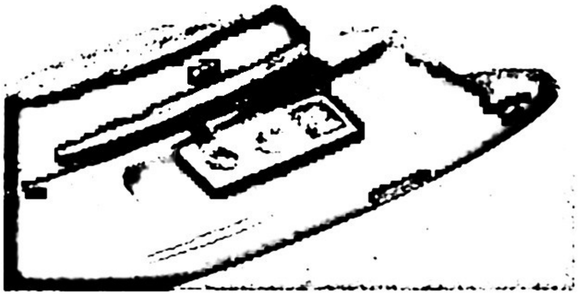


Figure 2 The wire that runs through the speaker system connects to two hook-up jacks on the driver.

So how does this fluctuation make the speaker coil move back and forth? The electromagnet is positioned in a constant magnetic field created by a permanent magnet. These two magnets -- the electromagnet and the permanent magnet -- interact with each other as any two magnets do. The positive end of the electromagnet is attracted to the negative pole of the permanent magnetic field, and the negative pole of the electromagnet is repelled by the permanent magnet's negative pole.

When the electromagnet's polar orientation switches, so does the direction of repulsion and attraction. In this way, the alternating current constantly reverses the magnet forces between the voice coil and the permanent magnet. This pushes the coil back and forth rapidly, like a piston. When the electrical current flowing through the voice coil changes direction, the coil's polar orientation reverses. This changes the magnetic forces between the voice coil and the permanent magnet, moving the coil and attached diaphragm back and forth.

When the coil moves, it pushes and pulls on the speaker cone. This vibrates the air in front of the speaker, creating sound waves. The electrical audio signal can also be interpreted as a wave. The frequency and amplitude of this wave, which represent the original sound wave, dictates the rate and distance that the voice coil moves. This, in turn, determines the frequency and amplitude of the sound waves produced by the diaphragm. Different driver sizes are better suited for certain frequency ranges. For this reason, loudspeaker units typically divide a wide frequency range among multiple drivers. In the next section, we'll find out how speakers divide up the frequency range, and we'll look at the main driver types used in loudspeakers.

2.3 Chunks of the Frequency Range

In the last section we saw that traditional speakers produce sound by pushing and pulling an electromagnet attached to a flexible cone. Although drivers all work on the same concept, there is actually a wide variety in driver size and power. The basic driver types are: 1) Woofers 2) Tweeters 3) Midrange (Figure 3).

Woofers are the biggest drivers, and are designed to produce low frequency sounds. Tweeters are much smaller units, designed to produce the highest frequencies. Midrange speakers produce a range of frequencies in the middle of the sound spectrum.

And if you think about it, this makes perfect sense. To create higher frequency waves -- waves in which the points of high pressure and low pressure are closer together -- the driver diaphragm must vibrate more quickly. This is harder to do with a large cone because of the mass of the cone. Conversely, it's harder to get a small driver to vibrate slowly enough to produce very low frequency sounds. It's more suited to rapid movement.

To produce quality sound over a wide frequency range more effectively, you can break the entire range into smaller chunks that are handled by specialized drivers. Quality loudspeakers will typically have a woofer, a tweeter and sometimes a midrange driver, all included in one enclosure.

Of course, to dedicate each driver to a particular frequency range, the speaker system first needs to break the audio signal into different pieces -- low frequency, high frequency and sometimes mid-range frequencies. This is the job of the speaker crossover.

Woofer Tweeter Midrange

Figure 3 Three basic driver types: woofer, tweeter, and midrange.

The most common type of crossover is passive, meaning it doesn't need an external power source because it is activated by the audio signal passing through it. This sort of crossover uses inductors, capacitors, and sometimes, other circuitry components. Capacitors and inductors only become good conductors under certain conditions. A crossover capacitor will conduct the current very well when the frequency exceeds a certain level, but will conduct poorly when the frequency is below that level. A crossover inductor acts in the reverse manner -- it is only a good conductor when the frequency is below a certain level (Figure 4).

When the electrical audio signal travels through the speaker wire to the speaker, it passes through the crossover units for each driver. To flow to the tweeter, the current will have to pass through a capacitor. So for the most part, the high frequency part of the signal will flow on to the tweeter voice coil. To flow to the woofer, the current passes through an inductor, so the driver will mainly respond to low frequencies. A crossover for the mid-range driver will conduct the current through a capacitor and an inductor, to set an upper and lower cutoff point.

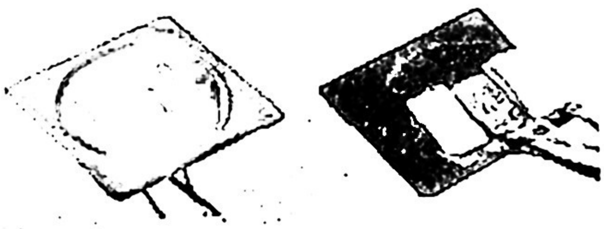


Figure 4 Typical crossover unit from a loudspeaker. The frequency is divided up by inductors and capacitors and then sent on to the woofer, tweeter and mid-range driver.

There are also active crossovers. Active crossovers are electronic devices that pick out the different frequency ranges in an audio signal before it goes on to the amplifier (you use an amplifier circuit for each driver). They have several advantages over passive crossovers, the main one being that you can easily adjust the frequency ranges. Passive crossover ranges are determined by the individual circuitry components -- to change them, you need to install new capacitors and inductors. Active crossovers aren't as widely used as passive crossovers, however, because the equipment is much more expensive and you need multiple amplifier outputs for your speakers.

Crossovers and drivers can be installed as separate components in a sound system, but most people end up buying speaker units that house the crossover and multiple drivers in one box. In the next section, we'll find out what these speaker enclosures do and how they affect the speaker's sound quality.

3. Description of the problem

The basic problem consists in the identification of sound signal quality. Of course, this requires a comparison between the real measured sound signal and the ideal good sound signal. We need to be able to accept speakers, which have a sound signal that do not differ much from the ideal signals. We show in Figure 5 (a) the form of the sound signal for a good speaker (of a specific type). The measured signal contains about 108 000 points in about 3 seconds. We need to compare any other measured signal with the good one and calculate the total difference between both of them, and if the difference is small then we can accept the speaker as a good one. On the other hand, if the difference is large then we reject the speaker as a bad one.

We show in Figure 5 (b) the sound signal for a speaker of bad quality. We can clearly see the difference in the geometrical form of this signal and the one shown in Figure 5 (a). In this case, the difference between the figures is sufficiently large and we easily determine that the speaker is of bad quality. We also show in Figure 5 (c) another sound signal for a bad quality speaker. Again, we can see clearly the difference in the form of the signal with respect to the good speaker.

(a) (b) (c)

Figure 5 Sound signal (a) Good Speaker (b) Bad Speaker (Case 1), (c) Bad Speaker (Case 2)

4. Fractal Dimension of a Geometrical Object

Recently, considerable progress has been made in understanding the complexity of an object through the application of fractal concepts [8] and dynamic scaling theory [11]. For example, financial time series show scaled properties suggesting a fractal structure [1, 2, 3]. The fractal dimension of a geometrical object can be defined as follows:

$$d = \lim_{r \to 0} [\ln N(r)] / [\ln(1/r)] \qquad (1)$$

where N(r) is the number of boxes covering the object and r is the size of the box. An approximation to the fractal dimension can be obtained by counting the number of boxes covering the boundary of the object for different r sizes and then performing a logarithmic regression to obtain d (box counting algorithm). In Figure 6 (a), we illustrate the box counting algorithm for a hypothetical curve C. Counting the number of boxes for different sizes of r and performing a logarithmic linear regression, we can estimate the box dimension of a geometrical object with the following equation:

$$\ln N(r) = \ln\beta - d \ln r \qquad (2)$$

this algorithm is illustrated in Figure 6 (b).

The fractal dimension can be used to characterize an arbitrary object. The reason for this is that the fractal dimension measures the geometrical complexity of objects. In this case, a time series can be classified by using the numeric value of the fractal dimension (d is between 1 and 2 because we are on the plane xy). The reasoning behind this classification scheme is that when the boundary is smooth the fractal dimension of the object will be close to one. On the other hand, when the boundary is rougher the fractal dimension will be close to a value of two.

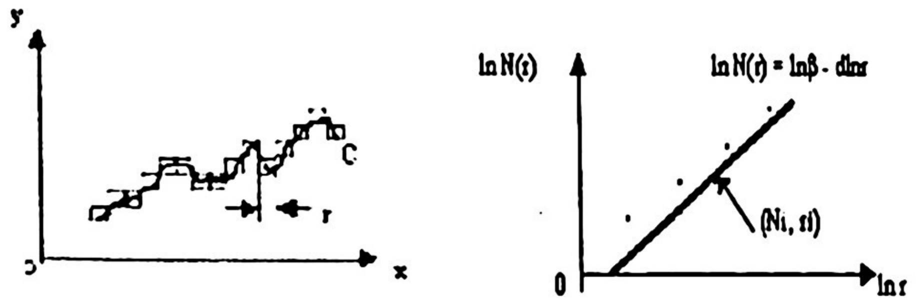


Figure 6 (a) Box counting algorithm for a curve C,
(b) Logarithmic regression to find dimension

We developed a computer program in MATLAB for calculating the fractal dimension of a sound signal. The computer program uses as input the figure of the signal and counts the number of boxes covering the object for different grid sizes. The fractal dimension for the sound signal of Figure 5(a) is of 1.6479, which is a low value because it corresponds to a good speaker. On the other hand, the fractal dimension for Figure 5 (b) is 1.7843, which is a high value (bad speaker). Also, for the case of Figure 5 (c) the dimension is 1.8030, which is even higher (again, a bad speaker).

5. Experimental Results

We describe in this section the experimental results obtained with the intelligent system for automated quality control. The intelligent system uses a fuzzy rule base to determine automatically the quality of sound in speakers. We used a neuro-fuzzy approach to adapt the parameters of the fuzzy system using real data from the problem. We used the time series of 108000 points measured from a good sound speaker (in a period of 3 seconds) as training data in the ANFIS approach. We then use the measured data of any other speaker as checking data, to compare the form of the sound signals. We show in Figures 7 and 8 two cases where the ANFIS approach is used to adapt a fuzzy system with training data of good sound speakers. The approximation is very good considering the complexity of the problem. Once the training was done, we used the fuzzy system for measuring the total difference between a given signal and the good ones. This difference is used to decide on the final quality of the speaker using another set of fuzzy rules with the Mamdani approach.

We show in Figure 9 (a) the implementation of the fuzzy rule base, in the Fuzzy Logic Toolbox of the MATLAB programming language. We also show in Figure 9 (b) the non-linear surface for modelling the quality control problem. Finally, we show in Figure 9 (c) the fuzzy rule viewer of MATLAB, in which the fuzzy rule base is used for specific values of the input variables.

The fuzzy rules are as follows:

IF Difference is small AND Fractal Dimension is small
THEN Quality is Excellent
IF Difference is regular AND Fractal Dimension is small
THEN Quality is Good
IF Difference is regular AND Fractal Dimension is high
THEN Quality is Medium
IF Difference is medium AND Fractal Dimension is small
THEN Quality is Medium
IF Difference is medium AND Fractal Dimension is high
THEN Quality is Bad
IF Difference is large AND Fractal Dimension is small
THEN Quality is Medium
IF Difference is large AND Fractal Dimension is high
THEN Quality is Bad
IF Difference is small AND Fractal Dimension is high
THEN Quality is Medium

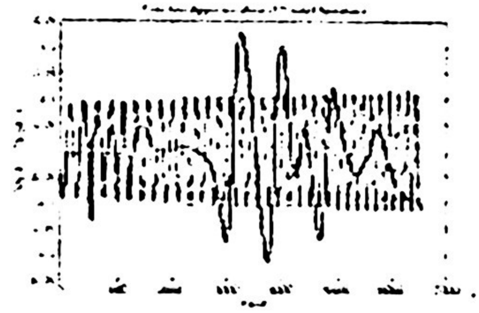


Figure 7 Function approximation of the sound signal using ANFIS (Case 1).

In the fuzzy rule base, the "Difference", "Fractal Dimension" and "Quality" are considered as linguistic variables in the fuzzy rules. We used gaussian membership functions for the linguistic values of all the variables. The membership functions for the linguistic values of these variables were tuned by using a simple genetic algorithm. Of course, the human experts on quality evaluation gave initial values for the fuzzy system, but these were optimized using real data for the problem and by applying the simple genetic algorithm.

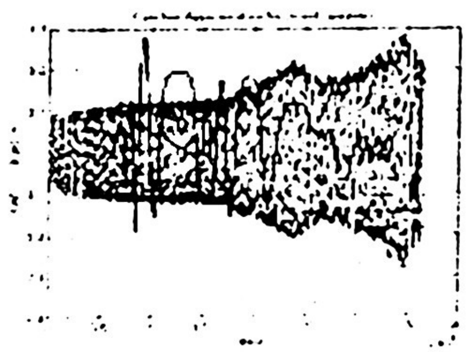
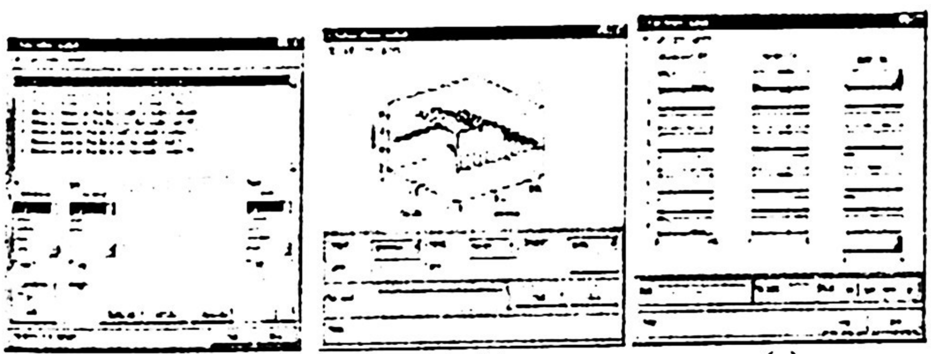


Figure 8 Function approx. of the sound signal using ANFIS (Case 2).



(a) (b) (c)

Figure 9 (a)Fuzzy rule base for quality control, (b) Non-linear surface for quality control, (c) Rule viewer for specific input values.

6. Conclusions

We described in this paper the application of a neuro-fuzzy-fractal approach to the problem of automating the quality control of sound speakers during manufacturing in a real plant. We have implemented an intelligent system for quality control in the MATLAB programming language using the ANFIS approach. We also use the fractal dimension as a measure of geometrical complexity of the sound signals. The intelligent system performs rather well considering the complexity of the problem. The intelligent system has been tested in a real manufacturing plant with very good results. We think that our approach for automating quality evaluation can used for similar problems with only minor changes in the membership functions and rules.

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