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A STUDY ON INTERFACE TECHNOLOGY BASED ON ANSYS AND PRO/E

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Abstract. This essay expounds the interface programming and data processing. It describes three interface forms of Pro/E and ANSYS: data file exchange, intermediate file exchange, and generic interface software exchange. The essay shows that generic interface software exchange is the most effective and best choice for ANSYS.

Keywords: Pro/E, ANSYS, interface, data, program

PRO/E and ANSYS Interface Form

Pro/E and ANSYS interface form is divided into three kinds: (1) Exchange data files directly. This approach is to set up a path that can be read directly in Pro/E model by using the method that supported in ANSYS. (2) Using intermediate files for data exchange. This approach is to derive Pro/E model for a special format, such as IGES into ANSYS, and then IGES files into ANSYS. (3) Convert by the interface software. Because of the parametric programming function of ANSYS, Pro/E identifiable data files can be converted into recognizable APDL program by the interface software.

Direct Exchange Of Universal Data Files. Direct exchange of universal data files is actually a kind of standard format conversion, and it is to use a standard data format file that has nothing to do with the system to implement the data exchange between multiple Pro/E systems. For example Pro/E can be saved as SAT [1], and ANSYS software can read the format of this file, in this way, ANSYS can share the data in the Pro/E. Not only in these two kinds of software in Pro/E, it can also save multiple formats, and other besides ANSYS CAE software also can read the same format. The versatility, simplicity, and standardized characteristics of this format make it widely used in integrated system format. This method has good data sharing characteristic, but if there are no some data in described format of Pro/E system in a standard data format, the product data will not be able to be completely "translated", and then result in the "lost" of data. At present common data interface standards have Parasolid format, IOES, STEP, STL, etc.

Parasolid format. It has strong modeling function, but it can only support a regular solid model. Its main functions include the mixed representation of free surface and analytical surface, entity operating a variety of ways, the topology and geometry data provide, etc. Parasolid CAD [2] is similar to that of ACIS data transfer and sharing standard, but there are different in terms of specific geometric modeling process.

IGES standard. The original developing of IGES is in order to make data exchange in the computer graphics database system. As IGES gradually mature, the standard has covered more
and more applications of CAD data exchange. Making IGES standard is to establish an information structure that used to define the purpose of the product data of digital expression and communication, as well as between different CAD system in the form of compatible product definition data exchange, so the IGES standard is the broader CAD standard, and not like ACIS and Parasolid that more limited to the geometric modeling techniques.

STEP standard. It is the 150 standard which describe how to express and exchange the information of product model data of digital product. Digital product data must include enough information to express the whole product life cycle which from design to analysis, manufacturing, quality control, testing, testing and production support functions. So STEP must cover the geometric topology assembly dimension tolerance constrained attributes and many other aspects of the content. TEP is constructed into a 150 standard which made up of many parts, and the basic part has been completed, most of others are in progress. One of the most important in STEP is scalability. Compared with the standard IGES, STEP standard has the following advantages: it formulated the corresponding application protocol for different fields, to solve the problem of adaptation of narrow IGES standard; Covered by this standard in addition to formal international standard including the control design of the project which has become in the field of 2 d figure 3 d configuration, and also includes the general mechanical design and process, electrician electrical and electronic engineering, shipbuilding, automobile manufacturing, etc.

The STL format. It is the combination of 3D model surface approximate that expressed for the small triangle plane. It is a kind of curved surface modeling, and it obey the vertex principle, orientation principle, total value and legal entity rules and so on. In describing aspects surface, compared with ACIS, IGES format, STL format has a big advantage, because this kind of expression method and finite element is similar to the pretreatment of surface mesh.

**Intermediate Files for Data Exchange.** This approach is to derived Pro/E model for a special format, such as IGES into ANSYS and then IGES files into ANSYS. This method is commonly used, and is easy to implement. Most of the CAD and CAE program retains the IGES interface, but as a result of the standard itself is not rigor, most complex models transferring end in failure.

**General Interface Software.** Due to the function of ANSYS with APDL parametric programming, the Pro/E identifiable data files can be converted into ANSYS APDL program through the interface software. This method can easily modify the model and mesh. Many of the CAD program have direct interface with ANSYS program, which written by ANSYS company or CAD software vendors. This kind of interface software is embedded in the application in a separate package. ANSYS also embedded the support for the Pro/E [3], and this is a Pro/E-ANSYS interface. It will not only convert Pro model data to ANSYS directly, but also provide the execution component based parametric design function. The function allows the components parameterized based on Pro/E model, using ANSYS program for optimization, and end with an optimization model, and still is based on components parameterized model. The module can provide engineers a good support when considering adopting what kind of post-treatment in the
process of finite element analysis. Using the software comes with interface can import the data quickly and accurately. Direct data transfer has some unique advantages, such as can modified model easily, divide grid for each individual, and all of these are very helpful for ANSYS post-processing. Convert the model data in this scheme is the most effective and optimal choice currently.

**B Interface**

**VB Features.** VB, because of its visualization, simplicity in the use and other aspects of the characteristics, is warmly welcomed by the majority of amateur programming enthusiasts. The latest VB6.0 has the following characteristics:

Visual programming: the so-called visual programming is that programmers do not have to write a lot of code in order design an interface, they can simply press the screen layout design according to the requirements, using the tools provided with the system [4] on the screen to draw a variety of "parts" that graphic objects, and set the properties of these graphical objects. VB will automatically generate interface design code, and programmers only need to write the implementation program function that part of the code, greatly improving the efficiency of programming. While traditional programming language in the design process, there is a need for all of the interface program code to be written, and in the process of preparing the practical effect of interface is not visible. Only to discover during operation, and must return to the program to go modify, edit. And many applications, programmers need to spend a lot of time and effort to debug the user interface, which not only increases the workload of programmers, but also the written interface lack of uniformity, versatility and practicality.

Object-oriented programming: VB adopted the object-oriented design ideas, the basic idea is to break down complex design issues can be completed independently of all functions - a relatively simple collection of objects. The so-called "object" is an operational entity such as forms, form of command buttons, labels, text boxes and other object-oriented programming sample like the building blocks can be in accordance with the requirements of interface design directly on the screen "painting" out of the windows, menus, buttons and so on the difference between object type and set properties for each object.

Structured programming: VB6.0 with structured programming statements[5], close to the natural language and human logic way of thinking, simple, and easy to understand, support color code, the editor can be automatically syntax error checking, and will have strong function and the use of flexible debuggers and compiler. In VB6.0 by use of program design process, can run the program at any time, and in the whole application design, can be compiled executable file, run directly in the WINDOWS environment.

Event-driven programming mechanism: run the Windows environment based on event-driven approach, each object can respond to multiple differences between events, each process events can drive the code - the code determines the object function is usually called the mechanisms
driving events. It can be triggered by user operation, and can also be triggered by system or application. Click the command button, for example, triggering the button Click event (Click), the event code will be executed. If the user did not do anything (not trigger events), you are in a wait state, and the application is composed of independent event process.

Support for multiple database system access: VB6.0 by use control system with data and database management window[6], can be directly set up or processing Microsoft Access and FoxPro database format, can also direct access to the Excel data in a spreadsheet. System also provides powerful data storage and retrieval functions. In your application, you can use structured query language (SQL) data standards, direct access to the database on the server.

Dynamic data exchange (DDE): using DDE communication need two Windows applications to the currently active server application sends a message to request information, and responsive server application, so as to realize data exchange between the two programs according to the information.

Object linking and embedding (OLE): VB6.0 by use of each application as an object, will link up different objects, then embedded in an application, thus can get a voice, video, images, animation, text and other information by collective files.

**Data Interface Program Feasibility Analysis.** Data interface program design flow chart 1 as shown, the entire process from start to finish the main part is the set of data analysis and read paragraphs title. And the HEADER section of the main content is the graphics associated with variable Settings, such as drawing boundaries, the range of entities, layer, etc. CLASSES part includes class defined in the application of the information; TABLE section contains four tables, each TABLE containing a variable number of list items. According to the linear table (LTYPE), the LAYER (LAYER), the words table (STYLE), depending on the chart (VIEW), etc. OBJECT part does not include the usually used to make the best of the information used to interact with other applications; BLOCKS period of record defines the first block of BLOCKS, the layer name, type of block, block of insert points and the block, all members of the block is divided into the kinds of graphics block [7], with attributes and anonymous BLOCKS. Anonymous blocks including with HATCH command generation section line and finish the dimensioning of with DIM command. The time of three tables and data extraction has less to do so in the process of reading data can be quickly skip this two parts in order to improve the running speed. And ENTITIES (ENTITIES) involves all the required data and information, point, line, plane coordinates, is the main part of the data analysis. Process can be according to the sentence reads:

```plaintext
Open dxf For lnPut As#l
Line InPut#l, n
ReDimx(n), y(n), z(n)
Fori=rlb
InPut#1, x(i), y(i), z(i),
NeXt
```
From 0 / SECTION / 2 / ENTITIES... / ENDSEC throughout the period of the end of 0, find out the logical relationship between pattern primitive, every record, each data is assigned to the corresponding variable address and preserved, do reserve for the after sequence translation. For this paragraph there are many kinds of functions and data processing method can be invoked, the effect of the different processing methods are also different.

**Interface Programming**

**Entity Segment Data Read.** Data files can be divided into different types according to different classification criteria. Data files can be divided into sequential files and random files according to data access methods and structures. The structure of sequential files is relatively simple. Records in the files stored one by one. In this document, only know the first record storage location but do not know the location of other records. When want to find a data, only from the beginning of the file and records can be read sequentially. That is to say, after reading record I, then can only read record I + 1, neither can read record I + 2 or I + 3, nor can read record I - 1 or I - 2, until find the record you want. The organization of Sequential files is relatively simple, as long as write the data records to a file one by one. But maintenance is relatively difficult. In order to change one record in the files, entire file must be read into the memory and re-written to the disk after modification.

In the random files, each record has a fixed length. The length of each field in the record also is fixed. In addition, each record of the random file has a number. While filling in, data will be put into the specified location directly as long as indicate the number. However while reading the data, the record can be read directly as long as give the number. In random files read and write operations can be performed simultaneously, which can find and modify each record quickly, need not to modify a record in the entire file to read and write operations. Random file data accesses flexible, convenient and faster than sequential file, but need larger space and more complex data organization. The DXF file, it is divided into header section (HEADER), the table segment (TABLE), class sections and physical segment blocks (ENTITIES) [8], but it can read the data only in sequential file style rather in random way. This is because although the structure of each segment is fixed, because of meta-data (including the number of points, lines, layers, etc.) or the operation sequence, file record length or position changes. For example, in entity segment, for the same graphics file, if change the drawing order, then the record of entity...
segment will change follow the drawing order. Therefore, this interface program read and writes DXF files in the form of a sequential file.

**Data Reading Program.** The operations to read and write Sequential files are in three steps: open the file; read the data file and close the file. Each step can be performed by different statements. To open the file in the program use statement “Open select file PathForInPutAs # l”.

To read data files using “LineInPut # l, ab” to achieve. However to close the file use “Close # l” operation. The difference to use different statements is not great, just according to personal habit.

The code of read Data is as follows:

```plaintext
Open selectfilepath For InPut As # 1
Do Until EOF (l)
    Line InPut # 1, ab
If ab = "ENTITIES" "Then
    For i = 0T020
        num = i
        Line InPut # 1, ab
    Next
    Line InPut # 1, xo (l)
    Line InPut # 1, y0 (l)
    Line InPut # 1, ab
    Line InPut # 1, 20 (l)
    Line InPut # 1, ab
    Line InPut # 1, xx (l)
    Line InPut # 1, ab
    Line InPut # 1, y0 (l)
    Line InPut # 1, ab
    Line InPut # 1, 22 (l)
    Line InPut # 1, ab
Forj = 1Tonum +5
    Line InPut # 1, ab
Nextj
Else
    Forj = 1Tonum a 1
        Line InPut # 1, ab
Nextj
End If
    Fork = 2To500
LineInPut # 1, a
    Line InPut # 1, xo (k)
    Line InPut # 1, ab
    Line InPut # 1, yo (k)
```


Line InPut # 1, ab
Line InPut # 1, 20 (k)
Line InPut # 1, ab
Line InPut # 1, xx (k)
Line InPut # 1, ab
Line InPut # 1, yy (k)
LineInPut # 1, ab
Line InPut # 1, ab
If ab = 210Then
For l = 1bnum +5
Line InPut # 1, ab
End If
LooP
Close # 1

Because hundreds of points are in DXF file. Line is formed by the connecting dots. For such vast amounts of data, the main function of this code is not only read to coordinate point data but also to capture logical information between graphics. Repeatedly used “for” loops and “if “conditional judgment statements judge the information in the codes. This code is mainly to read the information on the real segment, but other segments of data information is not what we need. So the cycle of physical segment begin with judge sentences in the program starts, and also use “ENDSEC” record to determine the end of the cycle, to maintain the high efficiency of the program.

**Data Processing**

**Data Coordinate Checking.** The data reading from DXF is a series of coordinates of the absolute coordinate. How to deal with the base coordinate directly as APDL data output will be very complicated. There may even be an error, and is not conducive to the verification by the user of module. The starting point In the coordinate data of DXF is not begin with O, but on the boundaries of any position in the CAD graphic[9]. For example, CAD drawing area is like a white paper. In the lower left corner of the white paper, it is the origin of the coordinate system. However, this white paper is large enough that the user cannot see the whole paper. What users can see is the part of the region by the display. When the user is drawing on the display area of this part, Point coordinates can be said to be random. But the dots are ideal relative relationship between the types of data.

**Accuracy of Data Processing.** In the process of program code writing and debugging, the coordinates of the graphics will be not in consistency to the designed data by the user. If users who wish to build a 10x20 rectangle, Ideally coordinate point is assumed to be the first point with coordinates (0,0,0),Coordinates of the second point (0,10,0), the coordinates of the third point (20,10,0), and the fourth coordinate points (20,0,0). However, in the Pro / E mapping process, as a general habit mapping project, there will be another set of coordinate’s case. The
first set of coordinates for the first point with coordinates (0,0,0), the coordinates of the second point (0.9,9.9999999999,0), the third point of coordinates (20,9.99999999,0), third the coordinates of point (20,9.99999999,0), the fourth point coordinates (20,0,0), Or the case of larger errors. For this phenomenon is not due to engineering personnel caused by carelessness, because the resulting visual reasons that a straight line is connected to the vertical and two points up. But it is not so that on the occasion. To address this situation, the following code can be used to solve it.

```vbnet
Public Function drawsingle(b As Double)
    C=1000000000000#
    If b*e-Int(b*e)>0.5 Then
        drawsingle=(hit(b*c)+l)/e
    Else
        drawsingle=(Ini(b*e))/e
    End If
End Function
```

The process of establishing a corresponding one module, it equivalents public function and can be called in a program of any events. This code will improve the accuracy of the data to the IE-12. If the accuracy is improved to IE-13, the problem is still not resolved. This is because that using VB to read DXF data can only save the data to the accuracy of IE-13. This error appears in mapping is very effective.

References


OPTIMIZATION OF FUEL CONTROL OF SUPERCRITICAL THERMAL POWER GENERATING UNIT

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Abstract. The general information and features of supercritical unit are illustrated. According to the engineering application situation, by using two feedforward methods, the application effect than conventional control methods, the response of unit is much faster than before. It’s benefit to Automatic Generation Control (AGC). It’s hoped that the analysis and optimization can provide useful experience and lessons for the design of the supercritical boiler’s fuel control to some extent.

Keywords: Supercritical unit; Feedforward; Optimization; Automatic generation control.

Introduction

During the past decade power generation has undergone several extremely significant changes. In today's power industry, a thermal power plant capable of making faster adjustments in power output in response to the system demand has significant competitive advantages. Such a plant may often be required to operate in a load-cycling or two-shifting manner resulting in non-linear changes in plant variables. A thermal power plant is a highly coupled large-scale multivariable dynamic system.

Although the new and effective theories and design methodologies being continually developed in the automatic control field, Proportional-Integral-Derivative (PID) controllers are still by far the most widely adopted controllers in industry owing to the advantageous cost/benefit ratio they are able to provide. In fact, although they are relatively simple to use, they are able to provide a satisfactory performance in many process control tasks. A thermal power is normally controlled by multi-loop PI/PID controllers. The control performance of these loops is adversely affected by inter-loop interactions. In addition, normal working of a power plant is severely affected by the occurrence of a range of system disturbances. Some common disturbances are changes in active burner configuration, heat-exchanger tube fouling, and variations in condenser vacuum. Being a highly coupled system, the disturbances in one part of the plant can have a significant effect on the rest of the plant as well.

In order to minimize the influence of both plant-wide interactions and disturbances so as to ensure a higher rate of load change without violating thermal constraints, a coordinated control strategy is required. But the reflect speed of boiler control is more slowly than turbine control, especially, mill control and all its issues. The time to fuel control from coal bunker to burning is
a quite long. So, to optimize mill control and all its issues, it is helpful to fit fuel control into the broader base of unit control.

**Unit Control System’s Structure**

The unit load controller essentially maintains the balance between thermal power in the boiler, and mechanical-electrical power developed by the turbine generator. Fundamental to this balance is the steam pressure at the inlet to the throttle valves (TVs) or turbine governor valves.

There are many ways in which this can be done, but increasingly coordinated or integrated controls are used as shown in Fig. 1. In this figure the controlled outputs are steam pressure and MW load, and the controlled variables are fuel flow and TV position. In operation the unit demand sets the set-point for pressure and power output, either locally or remotely from the load dispatch centre, and the control systems do the rest. With most plants now controlled by distributed control systems, it is fairly straightforward to set controller parameter values for stable operation over an acceptable load range. Variations of the controller structure are also possible, e.g. the use of derivative control in the feedforward signals.

There are some key issues that must be discussed in relationship to Fig. 1. The two most important issues are the use of pulverized fuel feedback in the fuel control loop and the contents of the milling group control. In practice, it is not possible to measure the pulverized fuel feedback and in addition the milling block is not just one mill but many mills. For example, it is up to six for a 660MW unit. This means that depending on load, mills must be switched in and out of service. It is important to mills that their mill capacity must fit to the requirement of unit load, so, the main steam pressure and temperature can keep stably.

**Fig. 1 Unit fuel and MW controllers of power plant.**
The Feedforward Control

Feedforward is a simple and powerful technique that complements feedback. Feedforward control provides a possibility to make control actions before disturbance response has occurred in the process output. Nowadays, Feedforward is implemented in most distributed control systems and the technique is used also for rather simple control problems to improve the control performance. The basic idea for design of the feedforward compensator is very simple. The ideal compensator is formed as the dynamics between the load disturbance and the process output divided by the dynamics between the control signal and the process output, with reversed sign. If the feedforward compensator is used, the effects of the load disturbance are eliminated from the process output.

Feedforward can be used both to improve the setpoint responses and to reduce the effect of measurable disturbances. So, feedforward mainly include two types, one is be made from disturbance, another is be made from setpoint.

Improved Setpoint Response. Feedforward can be used very effectively to improve the setpoint response of the system. By using feedforward it is also possible to separate the design problem into two parts. The feedback controller is first designed to give robustness and good disturbance rejection and the feedforward is then designed to give a good response to setpoint changes. Effective use of feedforward requires a system structure that has two degrees of freedom. It is first assumed that the system has the structure shown in Fig. 2. Let the process have the transfer function and a feedback controller $C(s)$. The feedforward compensator is characterized by the transfer functions $M_u(s)$ and $M_y(s)$.

The transfer function from setpoint to process output is

$$G_{ysp}(s) = \frac{P(CM_y + M_u)}{1 + PC} = M_y + \frac{PM_u - M_y}{1 + PC}$$

The first term represents the desired transfer function. The second term can be made small in two ways. Feedforward compensation can be used to make $PM_u - M_y$ small, or feedback compensation can be used to make the error small by making the loop gain $PC$ large. The condition for ideal feedforward is

$$M_y = PM_u$$
Notice the different character of feedback and feedforward. With feedforward it is attempted to match two transfer functions, and with feedback it is attempted to make the error small by dividing it by a large number. With a controller having integral action the loop gain is very large for small frequencies. It is thus sufficient to make sure that the condition for ideal feedforward holds at higher frequencies.

Theoretically and realistically, feedforward originate from setpoint can improve the reaction speed.

**Disturbance Attenuation.** Disturbances can be eliminated by feedback. With a feedback system it is, however, necessary that there be an error before the controller can take actions to eliminate disturbances. In some situations, it is possible to measure disturbances before they have influenced the processes. It is then natural to try to eliminate the effects of the disturbances before they have created control errors. The principle is illustrated in Fig. 3.

**Fig. 3 Block diagram of a system with a measured disturbance d**

Compared to Fig. 2, process transfer function $P(s)$ is composed of two factors, $P(s) = P_1(s)P_2(s)$. A measured disturbance $d$ enters at the input of process section $P_1(s)$. The measured disturbance is fed to the process input via the feedforward transfer function $G_{ff}(s)$. The transfer function from load disturbance to process output is

$$G_{yd}(s) = \frac{P_2(P_1 - P_1G_{ff})}{1 + PC} = P_2(P_3 - P_1G_{ff})S$$

Where $S = 1/(1 + PC)$ is the sensitivity function? This equation shows that there are two ways of reducing the disturbance. We can try to make $P_3 - P_1G_{ff}$ small by a proper choice of the feedforward transfer function $G_{ff}$, or we can make the loop transfer function $PC$ large by feedback. Feedforward and feedback can also be combined.

An ideal feedforward compensator is given by

$$G_{ff} = \frac{P_3}{P_1}$$
The ideal feedforward compensator is formed by taking the inverse of the process dynamics $P_1$. This inverse is often not realizable, but approximations have to be used.

Not only hydraulic, pneumatic actuator, but also electric actuators always don’t immediately act to reach the target, so the transfer function should consider the time problem. Additional, the actuator is proportional control. For example, process transfer functions are modeled as first-order systems with time delay, i.e.

$$P_1 = \frac{1}{10s + 1} e^{-2s}, \quad P_2 = 1, \quad P_3 = \frac{1}{5s + 1} e^{-2s}$$

Fig. 4 Schemes of mill capacity and primary air flow to mill

As shown in Fig. 4, $G_{ff}(s)$ is 0, $1, \frac{1}{5s + 1}$. Experimental results show that using the static-lag can enhance control system performance.

In a word, a feedforward controller can be very beneficial in solving the problem of achieving a satisfactory performance both in the setpoint following and in the load disturbance rejection task.

**Industrial Application**

As we known, primary air flow and temperature are significant influences in mill control. Secondary air flow is important in the furnace but does not affect the mill. But dynamics of the primary air response and the coal response should be considered. In practice, attempts are made using lag-lead filter networks.

**Control of Pulverized Fuel Flow.** Since pulverized fuel flow cannot be measured, it is usual to replace this measurement by the feeder speed measurement of fuel flow. In steady-state operation this is a satisfactory thing to do, but transiently there are significant differences resulting in challenging environmental problems during load change that significantly reduce maximum load change rates. Fig. 5 shows a more detailed description of the unit fuel control part of Fig. 1. Note that the fuel demand is for the entire mill group and this has to be split into the fuel demand for each of the individual mills.
It is very important that the milling group and hence the individual mills provide the correct amount of fuel, as set by the unit demand. For safe and efficient mill and furnace operation primary and secondary air flow must also be correct. Primary air flow and temperature are significant influences in mill control. Secondary air flow is important in the furnace but does not affect the mill. Its control is usually fairly simple and is done by measuring the air pressure in the hot air duct to the burners and controlling this by simple feedback to a desired set-point.

The basic idea behind the control of primary air and fuel to the mill, the sub-mill control system, is fairly straightforward and is based on the so-called 'load line' of the mill. This load line is predetermined for a mill and shows the relationship between the air mass flow and the coal mass flow required for the mill to operate in the safe air-fuel range. In order to ensure the air flow to meet the requirement of start-up of coal mill, the minimum air flow must be set up in advance. As shown in Fig. 6, is set by the need to establish a satisfactory recirculation load in the mill.

For example, one specific correspondence of a mill feeder and primary air flow is shown in Table 1. It is same to Fig. 6 that have a minimum air flow.
Table 1: Correspondence of a mill capacity and primary air flow to mill

<table>
<thead>
<tr>
<th>Feed Coal (t/h)</th>
<th>0</th>
<th>12.47</th>
<th>39.18</th>
<th>42.4</th>
<th>49.98</th>
</tr>
</thead>
<tbody>
<tr>
<td>Primary Air Flow (kg/s)</td>
<td>12.28</td>
<td>12.28</td>
<td>17.75</td>
<td>18.34</td>
<td>19.72</td>
</tr>
</tbody>
</table>

The air temperature is set by the requirements of having the coal sufficiently dry in the mill whilst at the same time not having the mill temperature too high and thus risking mill explosion.

![Graph showing trend chart]

1-coal; 2-primary air flow; 3-coal bunker temperature; 4-opening valve of cold primary air damper

Fig. 7 Actual application trend chart

**Application of Feedforward.** As we known, the open loop system has a faster response because it takes corrective action before the controller variable starts to change. When the open loop or feedforward system is combined with the closed loop or feedback system, the result is a fast response system that can compensate for change in the calibration curve.

The key issue of Automatic Generation Control (AGC) is the response speed of unit load, so, the unit load feedforward is necessary to get a quick response of the fuel control. Experiments proved to be effective that according to the range-ability of load (MW) to adjust fuel control in advance. Additional, if the pressure derivation is set reasonably as feedforward of fuel demand, it can shorten the responding period and ensure main steam pressure stable. Fig. 8 shows the feedforward design of fuel control.

As shown in Figure 9, design of fuel control is implemented with the feedforward dynamics compensation and the feedback PID control. Experimental results in a thermal power plant show that main steam pressure response speed and load follow characteristics have been greatly improved to meet the requirements of the power grid scheduling.
Steam Pressure
set-point
Steam Pressure
measurement

△

 PID
Pressure
controller

Fuel
demand

△
Generator MW
set-point
Generator MW
measurement

\( f_3(x) \) 
\( \sum \)

Fig. 8 Feedforward design of fuel control

1-fuel demand; 2-coal measurement;
3-main pressure setpoint; 4-main pressure measurement

Fig. 9 Example of unit control with feedforward

Conclusions

Two methodologies of feedforward have been thoroughly analysed in order to provide a clear characterization of them and to understand their applicability in practical situations. It has been shown that the use of such techniques indeed represents a valuable solution for the implementation of a high-performance controller by retaining at the same time the overall ease of use of a PID controller. A feedforward controller can be very beneficial in solving the problem of achieving a satisfactory performance both in the set-point following and in the load disturbance rejection task. In particular, the feedforward performances have been addressed. Features of the standard approach have been discussed. The selection of the correct parameters of feedforward in fuel control system may be a difficult and time consuming task. But the feedforward control action can be used to achieve a satisfactory performance.
References


TO USE SENSORY INFORMATION AS FEEDBACK FOR HIGH PERFORMANCE REAL-TIME ROBOTICS APPLICATIONS

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Abstract. The paper will demonstrate the techniques involved in balancing an unstable robotic platform. The platform will be an ideal for the implementations of both PID digital control and Kalman filter algorithms. Both algorithms will provide the necessary control for the system. Therefore the presented paper will investigate the performance of both PID digital control and Kalman filter algorithms. The control system performance is directly dependent on Kalman filter and PID controller input parameters. The results clearly show how the adjustable parameters on the control system directly affected the overall system performance.

Keywords: PID control, Kalman filter, inverted pendulum, feedback systems.

Introduction

“The research on balancing robots has gained momentum over the last decade in a number of robotics laboratories around the world”. The balancing robot platform technologies will eventually emerge as a new way of maneuverability and mobility in robotic applications. For example, a motorized wheelchair utilizing this technology would give the operator greater maneuverability and thus access to places most able-bodied people take for granted.

The presented research paper will investigate a two wheel balancing robot, which will be used as a test bed to examine the use of a digital control algorithm and a Kalman filter. The self balancing robot will be model after the inverted pendulum problem. The digital control algorithm that will be investigated as part of the self balancing robot is the PID controller [3]. The PID control algorithm will be used on the balancing robot to provide system stability. Another very important addition to the digital control system on the robot is the use of the Kalman filter. The filter is an estimation algorithm that is popular among the embedded control community. The Kalman filter is used to provide sensor fusion between the accelerometer and gyroscope. The digital filter will provide the reliable sensor data that will be used by the robot to get tilt angle information.

The method that will be used to control the self-balancing two wheeled robot will be a linear controller. It will be applied through a Proportional, Integral, and Derivative also refer to as the PID. The PID has proven to be popular among the control engineering community. As stated by Vance J. Van Doren, “For more than 60 years after the introduction of Proportional-Integral-Derivative controllers, remain the workhorse of industrial process control”. PID controller comprises of a proportional, an integral, and a derivative control part. Proportional Controller:
For many control applications, the abrupt change between a fixed motor control value and zero does not result in a smooth control behavior. We can improve this by using a linear or proportional term instead. The formula for the proportional controller (P controller) is \( R(t) = KP \cdot (v_{des}(t) - v_{act}(t)) \) The difference between the desired and actual speed is called the “error function”. Varying the “controller gain” KP will change the controller behavior. The higher the KP chosen, the faster the controller responds; however, a too high value will lead to an undesirable oscillating system. Therefore it is important to choose a value for KP that guarantees a fast response but does not lead the control system to overshoot too much or even oscillate.

Integral Controller : Unlike the P controller, the I controller (integral controller) is rarely used alone, but mostly in combination with the P or PD controller. The idea for the I controller is to reduce the steady-state error of the P controller. With an additional integral term, this steady-state error can be reduced to zero. When using \( e(t) \) as the error function, the formula for the PI controller is: \( R(t) = KP \cdot [ e(t) + \frac{1}{TI} \cdot \int_0^t e(t)dt ] \) We rewrite this formula by substituting \( QI = \frac{KP}{TI} \), so we receive independent additive terms for P and I: \( R(t) = KP \cdot e(t) + QI \cdot \int_0^t e(t)dt \).

Derivative Controller : Similar to the I controller, the D controller (derivative controller) is rarely used by itself, but mostly in combination with the P or PI controller. The idea for adding a derivative term is to speed up the P controller’s response to a change of input. The PD controller reaches equilibrium faster than the P controller, but still has a steady-state error. The full PID controller combines the advantages of PI and PD. It has a fast response and suffers no steady-state error [1].

The advantage of using the Kalman filter in the autonomous self balancing two wheel robot, that it can be used to provide a good estimate of vertical angle to control and maintain the robot balance. It can also be used to remove any measurement noise from the gyroscopes and accelerometers. A disadvantage to using the Kalman filter is that there is not a standard methodology or notation for the equations used for the filter; making the use of the filter more complex. As stated by both Greg Welch and Gary Bishop, “The Kalman filter is a set of mathematical equations that provides an efficient computational (recursive) means to estimate the state of a process, in a way that minimizes the mean of the squared error. The filter is very powerful in several aspects: it supports estimations of past, present, and even future states, and it can do so even when the precise nature of the modeled system is unknown.”

Architecture

The physical balancing robot is an inverted pendulum with two independently driven motors to allow for balancing, as well as driving straight and turning. The components used for making the robot are tilt sensors, inclinometer, accelerometers, gyroscopes, and digital cameras which are used for experimenting with this robot.

Gyroscope is a piezo-electric sensor designed for use in remote controlled vehicles, such as model helicopters. The gyroscope modifies a servo control signal by an amount proportional to its measure of angular velocity. Instead of using the gyro to control a servo, we read back the
modified servo signal to obtain a measurement of angular velocity. An estimate of angular displacement is obtained by integrating the velocity signal over time.

Acceleration sensors are used to output an analog signal, proportional to the acceleration in the direction of the sensor’s axis of sensitivity. Mounting two acceleration sensors at 90° angles means that we can measure the translational acceleration experienced by the sensors in the plane through which the robot moves. Since gravity provides a significant component of this acceleration, we are able to estimate the orientation of the robot.

Inclinometer is used to support the gyroscope. Although it cannot be used alone because of its time lag, it can be used to reset the software integration of the gyroscope data when the robot is close to resting in an upright position [7].

Digital camera can be used in two ways, either to support balancing or for detecting objects, walking paths, etc.

In order for the autonomous self balancing two wheel robot to remain in a vertical stable state, selection of good DC motors is important. DC motors with high torque output and fast RPM make them ideal to be used on a two wheel robot system. Selection of proper DC motors was an important great consideration for the robot system. The motors used for the autonomous self balancing two wheel robot are the popular Tamiya Gear head 380k75 DC motors.

A set of DC motor drivers are needed to support the Tamiya gear head 380k75 DC motors, to successfully work on the two wheel robot. The motor drivers are critical in getting any type motor to function properly. The drivers provide the high voltage and current levels outputs necessary to drive the motor. Some available motor drivers have inputs that allow the user to control the motors speed and direction. The VNH2SP30 is a dual DC motor driver that is able to drive two independent DC motors. Each DC motor driver has inputs that vary the motors speed and direction. The motor speed is controlled by a PWM input. Directions are determined by toggling the direction.

The power supply is another important component of the robot. There are many types of batteries that have different chemical makeup. Different battery types have advantages or disadvantages over one another in terms of power capacity. For the two wheel robot implementation a compact yet with high power capacity battery should be chosen. The battery type of choice is the nickel-metal Hydride batteries. Ni-MH batteries will provide the required power needed to drive all of the electrical devices on the two wheel robot.

The PID control strategy selected for implementation on the physical robot requires the measurement of four state variables: \{x, v, Θ, ω\}.
Table 1 State variables

<table>
<thead>
<tr>
<th>Variable</th>
<th>Description</th>
<th>Sensor</th>
</tr>
</thead>
<tbody>
<tr>
<td>x</td>
<td>Position</td>
<td>Shaft encoders</td>
</tr>
<tr>
<td>v</td>
<td>Velocity</td>
<td>Differentiated encoder reading</td>
</tr>
<tr>
<td>θ</td>
<td>Angle</td>
<td>Integrated gyroscope reading</td>
</tr>
<tr>
<td>ω</td>
<td>Angular velocity</td>
<td>Gyroscope</td>
</tr>
</tbody>
</table>

An implementation relying on the gyroscope alone does not completely solve the problem of balancing the physical robot, remaining balanced on average for 5–15 seconds before falling over. This can be an encouraging initial result, but it is still not a robust system. The system’s balancing can be greatly improved by adding an inclinometer to the robot. Although the robot was not able to balance with the inclinometer alone, because of inaccuracies and the time lag of the sensor, the combination of inclinometer and gyroscope proved to be the best solution. While the integrated data of the gyroscope gives accurate short-term orientation data, the inclinometer is used to recalibrate the robot’s orientation value as well as the gyroscope’s zero position at certain time intervals when the robot is moving at a low speed [4].

Methodology

The objective of the PID controller is to provide control and stability. The PID consists of three gain terms that affect the output response of the system. The proportional KP gain determines the rise time of the system. The integral KI gain affects the steady state error. The derivative KD gain controls the system response overshoot and helps with its stability.

The PID control algorithm is a very straightforward algorithm that provides the necessary output system response to control a process. One unique feature of the PID controller is that it is capable of manipulating the process inputs based on the history and rate of change of the signal. The
algorithm is best suited when the process under control is modeled as a linear system. This gives more accurate and stable control. The PID controllers are widely used on closed loop control systems, where the process output measurement is fed back to the system and gets processed by the controller.

The closed loop control system is also referred to as a negative feedback system. The basic idea of a negative feedback system is that it measures the process output from a sensor. The measured process output gets subtracted from the reference set-point value to produce an error. The error is then fed into the PID controller, where the error gets managed in three ways. The error will be used on the PID controller to execute the proportional term, integral term for reduction of steady state errors, and the derivative term to handle overshoots. After the PID algorithm processes the error, the controller produces a control signal. The PID control signal then gets fed into the process under control. The process under PID control is the two wheeled robot. The PID control signal will try to drive the process to the desired reference set point value. In the case of the two wheel robot, the desired set-point value is the zero degree vertical position [8].

In order for the closed loop system to work correctly, the PID controller must be correctly tuned. The constant values for the PID controller play a very vital role in the system output response. To find the PID constant gain values, a tuning procedure on the PID needs to be carried out. The sample time in the algorithm is also another important parameter when tuning the PID controller. There are numerous ways to tune the PID controller, to achieve desire output response. The widely used tuning algorithm is the Ziegler-Nichols which was introduced back in the 1940s. The Ziegler-Nichols PID tuning algorithm is the preferred choice to manually tune any PID controller used in industry today. To carry out the procedure to tune the PID controller using the Ziegler-Nichols is very straightforward. Since the self balancing two wheel robot is classified as a closed loop system. The close loop form of the Ziegler-Nichols tuning algorithm will be used to find the controller gain values. The sample rate, in a discrete PID control algorithm, plays an important role in the execution of the closed loop system. The sampling rate in a discrete time controller is both affected by the software and hardware structure of the controller. Choosing the right sampling rate in a discrete time controller determines its overall performance [6].

When using the PID control algorithm to control a system, the control engineering designer needs to be aware that not all parts of the system that need to be controlled need the full three term PID controller. If the system has a good output response with only the PI or PD controllers, there is no need to implement the full PID control algorithm. When using only the PI or PD controller, the implementation of control algorithm will be kept simple [2].

The Kalman is a recursive digital filter that provides a very effective means of estimating the state of any process. The Kalman filter can be used on a control system that is exposed to noisy environments because it minimizes the square error. The filter can reduce noisy measurements from sensors’ data before it’s fed into any control system. Before the Kalman filter can be used to get rid of noise from a sensor signal the process or system that is being measured must be modeled by linear system. For the autonomous self balancing two wheel robot, the Kalman filter
estimates the process by using a feedback scheme. First the filter estimates the system’s state at some time step and then gets noisy measurements in the form of feedback.

![Graph](image)

**Fig. 1**

In order for the Kalman filter to successfully be implemented onto the two wheel robot, an accurate model needs to be developed. The process also needs to be modeled as a linear system for Kalman filter implementation to be successful. The Kalman filter will use the data from the accelerometer to eliminate the drift problem from the gyroscope output signal. In the process of using the filter, unwanted noise from the accelerometer will either be minimized or completely eliminated. The Kalman filter process model will be modeled as a single dimensional inertial measurement unit. For the single dimensional inertial measurement unit, a two state Kalman filter is implemented to track the angle of the two wheeled robot and gyroscope bias value.

**Limitations**

A number of problems can be encountered with the sensors used. Over time, and especially in the first 15 minutes of operation, the observed “zero velocity” signal received from the gyroscope can deviate the desired result. This means that not only does our estimate of the angular velocity become inaccurate, but since our estimate of the angle is the integrated signal, it becomes inaccurate as well.

The control system assumes that it is possible to accurately generate a horizontal force using the robot’s motors. The force produced by the motors is related to the voltage applied, as well as the current shaft speed and friction.

In certain situations, the robot needs to generate considerable horizontal force to maintain balance. On some surfaces this force can exceed the frictional force between the robot tires and the ground. When this happens, the robot loses track of its displacement, and the control loop no
longer generates the correct output. This can be observed by sudden, unexpected changes in the robot displacement measurements [5].

The major constraint that the autonomous two wheel robot will have will be the sampling time that will be used to execute the control algorithms. The noisy measurements from the feedback sensors can impact the performance of control algorithm. Noisy sensor measurements give inaccurate results which will not allow PID control performance. Another major issue on the robot is the computation time that is used on the microcontroller to run the control algorithms. If there is an extensive delay, the robot will not be able to correct the angle in time to keep the robot stable.

**Conclusions**

Over the past few years, self-balancing robots have become a popular topic of research. Most of the research and development involves control algorithms and system dynamics. Advanced controllers provide robust and optimal control for self balancing robots. The two wheeled self-balancing robots are excellent examples of the classical inverted pendulum problem.

The experience of working on the classical problem of Inverted Pendulum is great. It is an ideal exercise to show one’s talent as Control Engineer. The work has gone a long way in helping us understand and develop an insight into the designing of control systems for control systems involving PID control. This paper provides a chance of designing a controller for a system that has a good dynamic behavior. The purpose of the project is to enhance the understanding of digital control algorithms and how the algorithms can be used to balance a robot on two wheels. The goal of the project is to provide the means of implementing a Kalman filter and PID controller.

**Recommendations**

This research paper provides the basis for future research on Kalman filter applications and implementations. Also to research more advanced control systems. The tuning of the Kalman filter needs to be studied more in depth. Tuning the Kalman filter by trial and error is time consuming. The linear PID control discussed in this paper was able to establish that the robot was able to balance under some minimal disturbances. Further fine tuning of the PID control algorithm is required for to achieve better output performance. Further research can be conducted to implement intelligent controllers on balancing robot system. Such intelligent controllers might be a fuzzy PID controller or a self tuning PID controller. Another future improvement to the self-balancing two wheel robot might be to add encoders. The encoders may be placed alongside the motors to provide motor rotation information. Having the motor rotation information from the encoders, the robot will have better locomotion control. Improved locomotion control will allow the robot to move precise distances. Using low cost components for the robot can sometimes lead to drawbacks, in this case from both the accelerometer and the gyroscope. Both sensors experience initial problems that made them difficult to us. The accelerometer data was
highly corrupted with unwanted noise. The gyroscope suffered from a problem that made its bias value drift away with time.

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YOUR WAVELET BASED PITCH DETECTION AND VOICED/UNVOICED DECISION

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Abstract. This paper describes property of the sudden change of a speech signal on its Glottal Closure Instant (GCI) and thereby discusses the principle of the localization of wavelets in both time and frequency domains. Based on this discussion, an algorithm for voiced/unvoiced segment decision and pitch detection is presented.

Keywords: speech processing, pitch detection, wavelet, voiced/unvoiced decision.

Introduction

Since the significant pitch value constructs the solid ground most speech coders are built on, pitch detection and voiced/unvoiced decision are the most crucial steps for the encoding and decoding process in the area of speech processing. Though many approaches involved in pitch detection have been proposed [1] [2], none of them has achieved a perfect result that can be applied under all the various circumstances with different requirements, such as men, women, children or singer, nor the one to be presented in this paper. This article is to come up with a wavelet-based way to accomplish pitch detection and voiced/unvoiced segment decision.

The normal speech segment can be divided into two categories, namely, voiced segment and unvoiced segment. The former comes from glottal vibration while the latter comes from airflow during phonation. The glottal vibration has the property of periodicity and the period of the vibration is called “pitch”, which is zero during unvoiced segment. It is known that the speech signal turns out to be a sudden change at the Glottal Closure Instant (GCI) [3][4], and as a result of this, the period between two GCIs can be detected to approximate the pitch of the corresponding voiced segment.

Singularity Detection By Wavelet

The main reason of using wavelet do detect pitch and determine voice from unvoice is due to its property of multi-resolution analysis and localization in both time and frequency domains.

If a real function \( \theta(x) \) satisfies

\[
\int_{-\infty}^{\infty} \theta(x) dx = 1,
\]

Where

\[
\theta(x) = O\left(\frac{1}{1 + x^2}\right),
\]
then it is called smooth function [5].

Since the energy of smooth function is mainly focused on low frequency, \( \theta(x) \) can be seen as an impulse response from a low-pass filter, and thus, to convolute a signal \( f(x) \) with \( \theta(x) \) can attenuate the high-frequency part of \( f(x) \) without changing its low-frequency part, and hence to make \( f(x) \) smooth.

Defining wavelet function \( \phi^1(x), \phi^2(x) \) following [5]:

\[
\phi^1(x) = \frac{d\theta(x)}{dx}, \quad \phi^2(x) = \frac{d^2\theta(x)}{dx^2},
\]

and let

\[
\theta_a = \frac{1}{a} \theta\left(\frac{x}{a}\right)
\]

then wavelet transform can be obtained from

\[
W_a^1 f(x) = f \ast \phi^1_a(x) = f \ast (a \frac{d\theta_a(x)}{dx}) = a \frac{d}{dx} (f \ast \theta_a(x))
\]

\[
W_a^2 f(x) = f \ast \phi^2_a(x) = f \ast (a^2 \frac{d^2\theta_a(x)}{dx^2}) = a^2 \frac{d^2}{dx^2} (f \ast \theta_a(x))
\]

It can be seen that \( W_a^1 f(x) \) and \( W_a^2 f(x) \) are direct ratio to the first and second order derivative of \( f \ast \theta_a(x) \) respectively, which obviously is the result from \( \theta_a(x) \) smoothing \( f(x) \). Since the extremum of the first-order derivative corresponds to the sudden change (singularity) in the original function, the point with maximum value of \( W_a^1 f(x) \) corresponds to the singularity of \( f(x) \). Consequently, if the first-order derivative of a smooth function is chosen as a wavelet function, the singularity point, e.g. \( x_0 \) of the original function \( f(x) \) can be detected by seeking the maximum of wavelet transform coefficients. Moreover, since \( W_a^1 f(x) \) always appears to be maximum at \( x_0 \) for several scaling factors \( a = 2^j \), the discontinuity of the original function \( f(x) \) will have transitivity under different resolutions [6].

This is the property that wavelets have localization in both time and frequency domain, which conforms perfectly to the requirement of speech pitch detection. When the Discrete Wavelet Transform (DWT) is used to detect the singularity within a speech signal, the period during the two singularities is just the pitch value. The scale factors \( a = 2^j \) indicate the degree of the localization of wavelets on both time and frequency domain. As the scale factor \( a \) increases, the frequency support of wavelet is shifted towards low frequency, the time resolution decreases but the frequency resolution increases.

Due to the localized feature of wavelets on both time and frequency domains, by means of different scale factors, the locations of the local maximums of DWT coefficients would specify the locations of GCI singularities and the pitch value accordingly.
Fig. 1 The algorithm of pitch detection and voiced/unvoiced segment decision.
The Algorithm For Pitch Detection and Voice/Unvoiced Decision

Fig. 1 illustrates the algorithm for pitch detection and voiced/unvoiced decision.

**Voiced/Unvoiced Decision.** Voiced/unvoiced speech segment is decided by the amplitudes of its DWT coefficients, as shown in Fig. 1. If the maximum value of the DWT coefficients is smaller than the T0, a threshold that has been set at the beginning of the program, then the segment being processed is determined to be unvoiced. If not, the segment may possibly be voiced and is left to be processed with pitch detection step, but if the pitch value detected turns out to be zero later on, this segment is still unvoiced, otherwise it is voiced. These two separate steps guarantee most of speech segments to be correctly decided concerning their voiced or unvoiced attributes. In this process, the most significant parameter is T0, whose appropriate value needs to be determined through constant attempts and adjustments during the practice and debugging process.

<table>
<thead>
<tr>
<th>n</th>
<th>h0(n)</th>
<th>h1(n)</th>
</tr>
</thead>
<tbody>
<tr>
<td>-3</td>
<td>0.0000</td>
<td>-0.00008</td>
</tr>
<tr>
<td>-2</td>
<td>0.0625</td>
<td>-0.01643</td>
</tr>
<tr>
<td>-1</td>
<td>0.2500</td>
<td>-0.10872</td>
</tr>
<tr>
<td>0</td>
<td>0.3750</td>
<td>-0.59261</td>
</tr>
<tr>
<td>1</td>
<td>0.2500</td>
<td>0.59261</td>
</tr>
<tr>
<td>2</td>
<td>0.0625</td>
<td>0.10872</td>
</tr>
<tr>
<td>3</td>
<td>0.0000</td>
<td>0.01643</td>
</tr>
<tr>
<td>4</td>
<td>0.0000</td>
<td>0.00008</td>
</tr>
</tbody>
</table>

Fig. 2. Filter coefficients of B-spline wavelet of order 3.

**Pitch Detection.** As shown in Fig. 1, pitch detection is more complicated and important compared with voiced/unvoiced decision. In the algorithm, the locations of local maximum coefficients calculated under different scaling factors need to be compared to give the specific locations of singularities as precise as possible, which determine the pitch value. The process is to find such pairs of singularities which are in two different scales respectively and have similar or the same location on horizontal axis (t axis). If any one location on the horizontal axis has two singularities in two different scales, it could be safely conferred that on this location a singularity is happening, i.e. a GCI. Then, the time distance between two closest pairs of such singularities is the pitch value.

Several wavelets were attempted in the algorithm, the “3th order B-spline wavelet” showed best performance in singularity detection. Therefore, it is selected for this algorithm to detect the singularities of a speech segment on its Glottal Closure Instants. Fig. 2 lists the sequence h0(n) and h1(n), i.e. the filter coefficients of B-spline wavelet of order 3. The interval between the two nearby GCIs is just the pitch period, and the frequency of normal human pitch is within the range from 30 to 500Hz.
“High” at different scales in one frame “Da Jia Dou Shuo Pu Tong Hua”

Fig. 3 demonstrates the wavelet transform coefficients in different scales of a speech segment sampled from an English word “High”. The algorithm in Fig. 1 was used, the frame length was set to 20ms and the sampling rate was 8K Hz. Considering d4 and d5 in Fig. 3, it can be seen that on the point of n=103, both of the two sequences reach one of their own local maximums, hence a singularity at this instant. Another singularity appears on n=132 in sequence d4 and n=138 in sequence d5, because the discrepancy of the two locations is 6, which is allowed by our experiment and could be deemed as a pair of similar locations. Then, the location of this singularity is (132+138)/2=135. Since there is no other singularities existing between these two, hence the pitch = (135-103)/8K = 4.0ms.

The flexible nature of this algorithm lies in the question that how close the two local maximums are required to be so that they can be seen as a pair of similar locations, i.e. to give a definition for “a pair of similar locations”. This usually has to be derived from repeated practices and also depends on the original signal, which means this can be different in the case of an adult’s speech and a child’s speech. This also gives rise to some wave glitches in the pitch outputs, as those in Fig. 4, which shows the raw output pitch by applying this algorithm on the original signal, a Chinese sentence “Da Jia Dou Shuo Pu Tong Hua”.

DWT Algorithm Applied
The DWT is achieved based on the famous Mallat Algorithm [7]. The Mallat Algorithm is an efficient fast Discrete Wavelet Transform (DWT) algorithm just like the status of FFT in the area of Fourier Transform. It is actually composed of two-channel filter banks in reality, and embodies fully the multi-resolution property of wavelets [8]. Fig. 5 illustrates the wavelet cascade decomposition hierarchically performed by Mallat Algorithm, where h0(n) represents
the Low-Pass Filter (LPF) and \( h_1(n) \) represents the High-Pass Filter (HPF). The coefficients and specifications of both filters depend on the type of wavelet selected.

![Diagram of the Mallat Algorithm]

\[ a = 2^j, j = 1 \]

\[ a = 2^j, j = 2 \]

\[ a = 2^j, j = 3 \]

**Fig. 5** The Mallat Algorithm.

![Diagram of the Atrous Algorithm]

\[ a = 2^j, j = 1 \]

\[ a = 2^j, j = 2 \]

\[ a = 2^j, j = 3 \]

\[ a = 2^j, j = 4 \]

**Fig. 6** The Algorithm Atrous

The Mallat Algorithm needs a subsampling operation made to the output from the filters at each scale in order to maintain the same two-channel filter bank to be used in the next scale. However, this leads to an undesirable condition in that the length of the output sequence is halved at the output of each scale. The critical requirement is then to make sure the wavelet transform sequences in different scales be in the same length. This means the extension of those sequences with coefficients to the original length needs to be done. The atrous algorithm [7] is therefore
introduced as a practical algorithm evolved from the Mallat algorithm. It shifts all the subsampling steps to the end of every route, as shown in Fig. 6.

In view of Fig. 5 and 6, the core algorithm to implement the wavelet decomposition can be summarized as:

\[
\begin{align*}
    j &= 0 \\
    a(j) &= x(n) \\
    \text{while } j < J \\
    d(j+1) &= a(j) * h1(j) \\
    a(j+1) &= a(j) * h0(j) \\
    \text{end of while}
\end{align*}
\]

where \(h1(j)\) and \(h0(j)\) are derived by interpolating \(2^j - 1\) zeros between any of the two values next to each other.

**Conclusions**

A wavelet-based algorithm is presented for pitch detection and voice/unvoice decision. The algorithm adopts 3th order B-spline wavelet rather than other wavelets that are more often employed, e.g. Daubechies wavelet family. Atrous Algorithm is used to make the wavelet transform sequences in different scales be in the same length. The result from this algorithm gives fresh and useful ideas and experience in wavelet-based pitch detection that can be left open and referred to during the future work in this field.

Wavelet algorithms will also be attempted on other audio coding steps. The strategy in deciding which two local maximums of two wavelet sequences are a pair of similar locations is simple and flexible. It also leads to glitches in the resulted pitch output, and therefore, it may still have the potential to be improved. For example, a smoothing process may apply.

**Acknowledgement**

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**References**


A FAIR SECURE MULTI-PARTY COIN-FLIPPING PROTOCOL

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Abstract. Designing coin-flipping protocol is one of the foundation problems in cryptography. 1986, Cleve showed that there doesn’t exist completely fair secure coin-flipping protocol when at least half of parties are malicious. From then on there were few papers discussed fairness (or some form of fairness) of coin-flipping protocol for more than two decades. In TCC 2009, based on the notion of 1/p-secure computation introduced by Katz, Moran et al. constructed an r-round 1/p-secure two-party coin-flipping protocol with bias \textit{O}(1/r). 2010, Beimel et al. extended Moran et al. results to multi-party model when less than 2/3 of the parties are malicious. In this paper, we propose another approach for coin-flipping protocol construction by using Garay et al.’s time-lines technique, and construct a fair secure multi-party coin-flipping protocol. Compared to the other coin-flipping protocols, our protocol enjoys two important advantages that its bias is 0 and there is no limitation of the malicious parties.

Keywords: coin-flipping, multi-party coin-flipping, time-lines.

Introduction

Coin-tossing protocols are protocols that generate a random bit with uniform distribution. Such a protocol should satisfy three properties. First, agreement is that when the protocol is run, even if some of the parties collude and deviate from the protocol's instructions, the honest parties output the same bit. Second, randomness is that the honest parties’ common output bit is uniformly distributed, i.e., the corrupt parties should not be able to bias the common output of the honest parties. Third, fairness is that either all the parties learn the output of coin-flipping functionality, or no party learns anything about the output. 1982, Blum[1] first constructed a two-party coin-flipping protocol which guarantees that the output of the honest party is unbiased only if the malicious party does not abort prematurely. This satisfies a rather weak notion of fairness in which once the malicious party is labeled as a “cheater” the honest party is allowed halt without outputting any value. While this notion suffices for some applications, in many cases fairness is required to hold even if one of the parties aborts prematurely. Then many researchers considered a stronger notion: even when the malicious party is labeled as a “cheater”, they require that the honest parties output a bit. We also discuss with this stronger notion of fairness in the paper. The latter notion of fairness turns out impossible to achieve in general. Specifically, Cleve[2] showed that for any two-party r-round coin-flipping protocol there exists an efficient adversary that can bias the output of the honest party by \textit{Ω}(1/r). From then on there were few papers discussed fairness (or some form of fairness) of coin-flipping protocol for more than two decades.
To achieve some form of fairness (strong notion), 2007, Katz [3] first introduced the notion of 1/p-secure computation. The definition uses the standard real/ideal simulation paradigm[4], except that he consider a completely fair ideal model (as typically considered in the setting of honest majority), and requires only 1/p-indistinguishability rather than indistinguishability. Based on this approach, 2009, Moran et al.[5] constructed an r-round 1/p-secure two-party coin-flipping protocol with bias $O(1/p)$. 2010, Beimel et al.[6] extended Moran et al. results to multi-party model when less than 2/3 of the parties are malicious. The same year, Gordon and J. Katz[7] first constructed a protocol for 1/p-secure two-party computation (2PC).

**Our Contribution.** We propose another approach for fair coin-flipping protocol construction by using a new technical tool---- Garay et al.’s time-lines[8], and construct a fair secure multi-party coin-flipping protocol. Compared to the other coin-flipping protocols, our protocol enjoys two important advantages that its bias is 0 and there is no limitation of number of the malicious parties.

**Organization.** We present the cryptographic tool----time-lines for our construction in Section 2. Then, in Section 3 we describe our multi-party coin-flipping protocol. Finally, in Section 4 we investigate our protocol, prove its security, and analyze its advantages.

**Time-Lines**

Gradual release timed commitments is a “timed cryptography” protocol, where the output is not revealed all at once. Rather, it is released gradually with the property that if an abort occurs, then the adversary has not learned much more about the output than the honest parties. In Crypto 2000, Boneh and Naor [9] first constructed a timed commitments scheme under generalized Blum-Blum-Shub (GBBS)[9] assumption. 2002, Garay and Jakobsson[10] introduced the notion of reusable time-lines which allow an amortization of the generation costs, and constructed time-lines by reusing the Boneh-Naor’s commitments. In their construction, assuming one time-line $L$ is already known, one can easily derive a new time-line from $L$, by raising the seed and every point in $L$ to a fixed power $\alpha$. Clearly, the result is a time-line with the same modulus. 2006, Garay et al. [8] gave a more efficient time-lines, which will be used to construct our multi-party coin-flipping protocol. 2003, Pinkas[11] first presented a fair secure 2PC protocol based on timed release commitment. 2008, Kiraz and Schoenmakers[12] improved upon Pinkas construction and proved its security according to the standard simulation paradigm.

The following is the details of Garay et al.’s time-lines.

**Setup:** Let $n$ be a positive integer representing a certain security parameter, and $\kappa$ be another positive integer representing the time-lines security parameter. For 80-bit security, one can take $\kappa = 80$.

1. Compute the master time-line
   A trusted third party run the following steps:
(1) Generate two random n-bit safe primes $p_1$ and $p_2$ such that $p_1=p_2 = 3 \mod 4$ and $p_1=(p_1-1)/2, p_2=(p_2-1)/2$ are also primes, and computes the safe Blum integer $N = p_1 \cdot p_2$. He also picks a random $g \in Z_N^*$.

(2) Compute $u_i = g^{2^i - 2^k} \mod N = \text{ReqSqu}_{N,g}(2^k - 2^{i-1})$ (Let $\text{ReqSqu}_{N,g}(x)$ denote $g^x \mod N$) for $0 \leq i \leq \kappa$. This computation can be performed by computing $a_i = 2^i \mod \varphi(N)$ and then $u_i = g^{a_i} \mod N$.

(3) Generate a proof that $u_i (0 \leq i \leq \kappa)$ are well-formedness. For details of the proof, we refer to [9].

(4) Output the master time-line $L=(N, g, \{u_i\}_1^\kappa)$, along with the above proof, which we assume is non-interactive.

2. Compute the derived time-line

(1) Pick a random value $\alpha \in [0, N/4]$.

(2) Set $h = g^\alpha$, and compute a derived time-line $L'=(N, h, \tilde{u})$, where $v[i] = (u[i])^\alpha$ for $0 \leq i \leq \kappa$.

**Lemma 1 (Strong Pseudorandomness)**[8] : Let $L=(N, g, \tilde{u})$, $L'=(N, h, \tilde{v})$ and $\kappa$ be as above protocol. Let $\delta$ be as in the GBBS assumption. Let $\tilde{w}$ be the vector containing the last $(l + 1)$ elements in $\tilde{v}$, i.e., $\tilde{w} = (v[\kappa - l], v[\kappa - l + 1], ..., v[\kappa])$. Let $A$ be a algorithm whose running time is bounded by $\delta \cdot 2^l$ and $R$ be a random element in $Z_N^*$. Then, assuming the composite decisional Diffie-Hellman assumption(CDDH)[13] and GBBS hold, there exists a negligible function $\varepsilon(\cdot)$ such that, for any $A$,

$$\Pr[A(N, g, \tilde{u}, h, \tilde{w}, v[\kappa - l - 1]) = 1] - \Pr[A(N, g, \tilde{u}, h, \tilde{w}, R^2) = 1] \leq \varepsilon(\kappa).$$

**A New Protocol for Multi-party coin-flipping Functionality**

Based on Garay et al.’s time-lines, we construct a new fair protocol, PmCoinFlip, which securely realizes multi-party coin-flipping functionality in the $(F_{2^k}^{\text{DL}}, F_{2^k}^{\text{DH}}, F_{2^k}^{\hat{R}})$-hybrid model. Turning to our protocol construction, we need the “one-to-many” $F_{2^k}$ functionality (see Figure 1) for the following three relations.

(1) Discrete log: $\text{DL} = \{(N, g, h, \alpha) \mid h = g^\alpha \mod N\}$

(2) Diffie-hellman quadruple: $\text{DH} = \{(N, g, h, x, y, \alpha) \mid h = g^\alpha \mod N \land y = x^\alpha \mod N\}$

(3) Blinded relations: Given a binary relation $R(y, x)$, we define a “blinded” relation $\hat{R}$ as follows:

$$\hat{R}(N, g, h, w, z, y, \alpha) = (h = g^\alpha \mod N) \land R(y, \frac{z}{w^\alpha} \mod N).$$
Functionality $F_{zk}^R$

$F_{zk}^R$ proceeds as follows, running parties $P_1, \ldots, P_n$, and an adversary $S$:

- Upon receiving $(zk\text{-prove}, sid, ssid, x, w)$ from $P_i$: If $R(x, w)$ does not hold, ignore. Otherwise, request $S$ for permission to send $(ZK\text{-PROOF}, sid, ssid, P_i, x)$ to each of $P_j (j \neq i)$. Send the messages as permissions are granted.

Fig. 1 The (multi-session) zero-knowledge functionality for relation $R$.

Our protocol consists of four phases: commit phase, prove phase, open phase, and output phase. Each party picks a random $h$-bit $x_i$ and commits to it in the commit phase. Proves correctness of the committed value in the prove phase, then open it gradually in the open phase, finally in the output phase, performs the following steps:

1. Broadcast $(h_i, c_i, y_i)$ be able to $P_i$ and check open.

2. All parties broadcast $(h_i, \delta y_i)$ in $\hat{\delta}$. Each party behaves differently in the first $m-1$ rounds from the last $\kappa-\mid m+1 \rounds$. At round $m$ of the open phase, the parties switch its strategy. From this round on the adversary may be able to force open the commitment. So the parties should find the openings in this round.

The details of our protocol are as following.

**Setup:** Compute the master time-line $L=(N, g, \bar{u} = \{u_i\})$ as the previous section time-lines.

**Round 1 (Commit phase).** For each party $P_i$, $1 \leq i \leq n$, performs the following steps:

1. Choose a random value $\gamma_i \in [1, \frac{N-1}{2}]$, set $h_i = g^{\gamma_i}$, and compute a derived time-line $L_i=(N, h_i, \bar{v}_i)$, where $v[j] = u[j]^\gamma_i$ for $1 \leq j \leq \kappa$.

2. Pick a random $h$-bit $x_i \in [1, N-1]$ and commit to it, $c_i = x_i \cdot v[k] = x_i \cdot (u[k])^\gamma_i$. Broadcast message (COMMIT, sid, $P_i, h_i, c_i$).

3. Send message $(zk\text{-prove}, sid, 0, P_i, (N, g, h_i, v[i], \gamma_i))$ to the $F_{zk}^{dl}$ functionality. All parties broadcast (RECEIPT, sid, $P_i$) after receiving messages (ZK-PROOF, sid, 0, $P_i, (N, g, h_i)$) from $F_{zk}^{dl}$, otherwise, set $x_i = \bar{x}_i$ ( $\bar{x}_i$ is the default value of party $P_i$) and continue.

**Round 2 (Prove phase).** For each party $P_i$ who has finished the commit phase successfully, does:

1. Send message $(zk\text{-prove}, sid, 0, P_i, (N, g, h_i, u[k_i], c_i, y_i), \gamma_i)$ to the $F_{zk}^k$ functionality.

2. All parties broadcast (PROOF, sid, $P_i, y_i$) after receiving messages (ZK-PROOF, sid, 0, $P_i, (N, g, h_i, u[k_i], c_i, y_i)$) from $F_{zk}^k$, otherwise, set $x_i = \bar{x}_i$ ( $\bar{x}_i$ is the default value of party $P_i$) and continue.

**Round $r=3, \ldots, \kappa+2$ (Open phase).** Let $l = r-2$. For each party $P_i$ who has finished the commit and prove phase successfully, does:
(1) Broadcast (RELEASE, sid, v[i]) and send message (zk-prove, sid, r, (N, g, hi, u[l], v[i]), \(y_i\)) to the functionality \(F_{zk}^m\).

(2) After receiving all n RELEASE and ZK-PROOF messages, proceed to the next round. Otherwise, if any party’s (assume it’s \(P_k\)) broadcast messages is missing, go to panic mode.

At the end of round \((\kappa + 2)\), compute \(x_j = c_j \cdot (v_j^{[\kappa]} \cdot r)^{-1}\) for \(1 \leq j \leq n\), output (DATA, sid, \(x_1, \ldots, x_n\)) and terminate.

**Panic mode:** For each party \(P_i, 1 \leq i \leq n\), does:
- If \(r \geq m\), use \(v_j^{[l-1]}(j = 1, \ldots, n)\) from the previous round to compute \(v_j^{[\kappa]}\) as \(v_j^{[\kappa]} = \text{RepSq}_{N, v, [v]}(2^{k-1} - 1)\), then computer \(x_j = c_j \cdot (v_j^{[\kappa]} \cdot r)^{-1}\). Output (DATA, sid, \(x_1, \ldots, x_n\)) in round \((\kappa + 2)\) and terminate.
- Otherwise, set \(x_i = \overline{x_i}\) (\(\overline{x_i}\) is the default value of party \(P_k\)) and continue the open phase.

**Round \(\kappa + 2\) (Output phase).** For each party \(P_i, 1 \leq i \leq n\), computes \(b_j = x_j^i \oplus x_j^2 \oplus \ldots \oplus x_j^n\) for \(1 \leq j \leq n\), and \(c = b_1 \oplus b_2 \oplus \ldots \oplus b_n\), then outputs \(c\).

This finishes the description of PmCoinFlip protocol.

### Analysis of Protocol PmCoinFlip

**Security Analysis.** Compared to the other coin-flipping protocols, our protocol enjoys an important advantage that its bias is 0.

**Claim 1.** In protocol PmCoinFlip, there exists no efficient adversary \(A\) that can bias the honest parties’ output with probability at least \(\mu(n)\) (\(\mu(n)\) is a negligible function in which for every positive polynomial \(p(n)\) there exists an integer \(N\) such that for all \(n > N\),

\[
\mu(n) < \frac{1}{p(n)}.
\]

**Proof.** If a party aborts at the \(r(\geq m)\) round, all the parties are able to compute the output bit \(c\) by forcing open the commitments. So, we only consider the adversary \(A\) corrupting the parties before the \(m\) round in which case their default values will be used to compute the output bit. Assume the first \(j\) parties are corrupted before the \(m\) round.

In this case we get \(b_i = x_i^1 \oplus x_i^2 \oplus \ldots \oplus x_i^n\) for \(1 \leq i \leq j\) and \(b_j = x_j^1 \oplus x_j^2 \oplus \ldots \oplus x_j^n\) for \(j + 1 \leq i \leq n\), and

\[
c_{\mu} = b_1 \oplus b_2 \oplus \ldots \oplus b_j \oplus b_{j+1} \oplus \ldots \oplus b_n.
\]

Let \(b_{\mu} = \overline{b_1} \oplus \overline{b_2} \oplus \ldots \oplus \overline{b_j}\) and \(b_{\mu} = b_{j+1} \oplus \ldots \oplus b_n\), then \(c_{\mu} = \overline{b_{\mu}} \oplus b_{\mu}\). Therefore,

\[
\Pr(c_{\mu} = 1) = \Pr(b_{\mu} = 1|\overline{b_{\mu}} = 0) + \Pr(b_{\mu} = 0|\overline{b_{\mu}} = 1)
\]

\[
= \Pr(b_{\mu} = 1) \cdot \Pr(\overline{b_{\mu}} = 0) + \Pr(b_{\mu} = 0) \cdot \Pr(\overline{b_{\mu}} = 1)
\]

\[
= \frac{1}{2} \cdot \Pr(\overline{b_{\mu}} = 0) + \frac{1}{2} \cdot \Pr(\overline{b_{\mu}} = 1)
\]

\[
= \frac{1}{2}.
\]
\[ \frac{1}{2} \cdot (\Pr(\bar{b}_1 = 0) + \Pr(\bar{b}_1 = 1)) \]
\[ = \frac{1}{2}. \]

In the above computation, (1) comes from independence of \( b_i \) and \( \bar{b}_i \), and (2) from the randomness of \( x_i \) (\( j + 1 \leq i \leq n \)).

Thus, we learn that there doesn’t exist an efficient adversary \( A \) that can bias the honest parties’ output with probability at least \( \mu(n) \).

Theorem 1. Under CDDH and GBBS assumption, the protocol \( PmCoinFlip \) securely realizes multi-party coin-flipping functionality with \( O(n) \)--investment in the \( (F^{DL}_Z, F^{DH}_Z, F^{k}_Z) \)--hybrid model.

**Proof.**

(1) **Agreement.** From the description of protocol \( PmCoinFlip \), we can easily get the agreement property.

(2) **Randomness.** From Claim 1, we learn the randomness property.

(3) **Fairness.** We will analyse fairness in the following two cases.

If a party aborts before the \( m \) round, from Lemma 1, he can’t compute the committed values \( x_1, \ldots, x_n \) by forcing open the commitments, then he couldn’t learn the output bit \( c \), the same as the other parties. In this case, it’s fairness that no party gets the output bit \( c \).

If a party aborts at the \( r (r \geq m) \) round where he is able to force open the commitment, the other parties switches its strategy and compute the output bit \( c \) by forcing open the commitments. In this case, it’s also fairness that all the parties learn the output bit \( c \).

From the above 3 steps, we get that protocol \( PmCoinFlip \) securely realizes multi-party coin-flipping functionality.

**B. Others Analysis.** We will analyse our protocols by comparing to the representative Beimel et al.’s [6] multi-party coin-flipping protocol (denoted as Bei10). We discuss the following three properties: bias, limitation of number of the malicious parties, and whether it’s according to the standard simulation paradigm.

Our results are summarized by the following table 1.

<table>
<thead>
<tr>
<th>Scheme</th>
<th>Bias</th>
<th>Number of malicious parties</th>
<th>According to the standard simulation paradigm?</th>
</tr>
</thead>
<tbody>
<tr>
<td>Bei10</td>
<td>( O(1/r) )</td>
<td>( \leq 2/3 )</td>
<td>Yes</td>
</tr>
<tr>
<td>Ours(news)</td>
<td>0</td>
<td>No limitation</td>
<td>No</td>
</tr>
</tbody>
</table>

From the above table, we know that our protocol has advantages on the bias and limitation of number of the malicious parties, but it isn’t according to the standard simulation paradigm.
Conclusion and Future Work

We introduced Garay et al.’s “time-lines” approach for fair secure coin-flipping protocol construction, and proposed a fair secure multi-party coin-flipping protocol. Compared to the other coin-flipping protocols, our protocol enjoys two important advantages that its bias is 0 and there is no limitation of number of the malicious parties. But our protocol wasn’t according to the standard simulation paradigm. It would be interesting to extend our approach to the standard simulation paradigm.

References


READOUT INTEGRATED CIRCUIT DESIGN FOR INFRARED DETECTOR WITH DUAL SWITCHED TYPE
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Abstract. This research designed a readout circuit with dual type switching for infrared sensors. The pixel readout circuit structure uses DI (direct injection) and CTIA (capacitive transimpedance amplifier). The readout circuit has four operating modes, which control single or dual band using signal control and the TSMC 0.35um 2P4M 5V process. The input current middle wavelength is set between 2nA to 4nA, the long wavelength is set between at 6nA to 8nA, the output voltage swing is 2.8V, and power consumption is 22.19mW

Keywords: Direct Injection, Capacitive Transimpedance Amplifier, Readout Circuit.

Introduction
Research related to infrared imaging sensors in recent years compares the previous single-band infrared sensors readout circuit, which can detect a specific range of the infrared band [1,2]. The advantages of the dual-band infrared sensor readout circuit are improved environment tolerance, enhanced video signal, increased dynamic tracking range, and better contrast and brightness. Infrared sensors are susceptible to environmental noise impact of temperature, which results in resolution attenuation. The dual-band infrared sensor read circuit was developed by enhancing resolution, thus, it receives signals from a larger range and improves environmental noise tolerance to obtain the best signal.

This paper presents a 40x16 dual-band readout circuit with dual type switching for infrared sensors. The switch mode is used to achieve two types of pixel circuit transformation. The readout circuit has four operating modes due to the additional control signal. The first part of this paper provides a detailed description of the readout circuit structure and operation principles, the second part presents the results of the readout circuit simulation, while the final part offers conclusions and discussion for the circuit design.

Design of Dual Type Switching Readout Circuit
A. System Structure. The structure of the 40x16 infrared readout circuit system blocks is shown in Figure 1. The circuit system includes a switch for a dual type pixel circuit, a column stage circuit, a bias circuit, and an output stage circuit. The system structure is applicable to a single band or a dual band sensor. The middle wavelength of the dual band simulation input current setting is set between 2nA to 4nA, while the long wavelength is set between 6nA to 8nA. The 40x16 readout circuit is column interlaced.
B. Switch of Dual Type Pixel Readout Circuit. The structure of the dual type switching pixel readout circuit is as shown in Figure 2. Inside the unit pixel, there are two types of pixel circuit transformers, DI and CTIA[3,4], which are used with switches S1 and S2, respectively. When switch S1 is closed, the pixel readout circuit type is DI. The first circuit operated is reset by the INTR1 of MCN2. The charge is clear for CINT1. When the circuit is reset, it creates a fixed bias voltage for the photo current, which is sensed by the sensor as injected through MCp1, and the integration capacitor obtains the integration voltage. The current charge transfers to voltage according to equation (3.1). Obtain sampling voltage VSH by means of MCN3’s sampling. Finally, the signal selected by the ROW of MCp8 and the signal passes to the next stage for output.

When switch S2 is closed, the pixel readout circuit type is CTIA, and the first operated circuit is reset by INTR2 of MCp5. The charge is clear for CINT2. When the circuit is reset, the sensed photo current is injected through MCp5, and the integration capacitor obtains the integration voltage. The operating signal mode is sampled and passes to the next stage, as in the upside depiction

\[ V_{\text{int}} = \frac{I \times t}{C_{\text{int}}} \]
Simulation Result

The dual band readout circuit output simulation middle wavelength current is set between 2nA and 4nA, and the integration time is 250μs, while the long wavelength current is set between 6nA and 8nA, and integration time is 125μs. The readout circuit has four operating modes.

Fig. 3 (a) shows the results of T1=0 and T2=0 mode simulations for the dual band DI mode. Fig. 3 (b) shows the results of T1=1 and T2=1 mode simulations for the circuit operations of the dual band CTIA mode. Fig. 3 (c) shows the results of T1=0 and T2=1 mode simulations of the circuit operations for dual band CTIA and DI modes. Fig. 3 (d) shows the results of T1=1 and T2=0 mode simulations of the circuit operations for dual band DI and CTIA modes. Table 1 shows the Simulation result of dual type switching readout circuit, which operates at room temperature, with output swing of 2.8V, and power consumption of 22.19 mW.

<table>
<thead>
<tr>
<th>Mode</th>
<th>Voltage (V)</th>
</tr>
</thead>
<tbody>
<tr>
<td>T1=0, T2=0</td>
<td>2.8</td>
</tr>
<tr>
<td>T1=1, T2=1</td>
<td>2.8</td>
</tr>
<tr>
<td>T1=0, T2=1</td>
<td>2.8</td>
</tr>
<tr>
<td>T1=1, T2=0</td>
<td>2.8</td>
</tr>
</tbody>
</table>

Fig. 3 Simulation results of readout circuit with four operate modes.
Table 1 Specifications for switch of dual type readout circuit

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Specification</th>
</tr>
</thead>
<tbody>
<tr>
<td>Readout pixels</td>
<td>40×16 pixels</td>
</tr>
<tr>
<td>Technology</td>
<td>TSMC 0.35μm 5V 2P4M</td>
</tr>
<tr>
<td>Input configuration</td>
<td>DI, CTIA</td>
</tr>
<tr>
<td>Power supply</td>
<td>5v</td>
</tr>
<tr>
<td>Maximum clock rate</td>
<td>3MHz</td>
</tr>
<tr>
<td>Output swing</td>
<td>1.6~4.4 v(2.8v)</td>
</tr>
<tr>
<td>Power dissipation</td>
<td>22.19 mW</td>
</tr>
<tr>
<td>Input photo current</td>
<td>MWIR: 2.nA<del>4 nA; LWIR: 6nA</del>8nA</td>
</tr>
</tbody>
</table>

Conclusions

This study designed a single and dual band infrared sensor read circuit. The dual switching type readout circuit is used in the structure of a dual band read circuit, and includes two pixel control circuits, resulting in middle and long band circuits with independent controls to adjust integration time. The dual type switching readout circuit is selected for different types of pixel circuit architectures according to the switching circuit suitable for different sensor bias.

Acknowledgement

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References


DESIGN OF AUTOMATION SYSTEM BASED ON EMERGING TECHNOLOGY

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Abstract. To smash the walls between industries, this article designed a general automation system on the bases of the alikeness of the automation system. The automation system fused the demands of industrial automation, electric power automation, building automation, enterprise information automation, etc. The automation system took full consideration of facts of the advanced management of statute, data compression and storage management, graph visual design, etc. It would help to the construction investment of compression system, reduce the difficulty of system maintenance, and improve the practical aesthetics system.

Keywords: Automation System, Compress and Access.

Introduction

We will find that all the industries have their own automation system. The designers of these systems often only focus on their own field, and know little about the systems of other fields. For example, MS information acquisition and processing are not easy in the railway monitoring system, but in electric power industry, such techniques have already developed, which can provide millisecond current information and precise information synchronization [1]. Therefore, the integration of a variety of technologies will be able to promote a change of the automation system.

The age of the Internet provides very good information channel for us, the special technology for a special industry or field, will gradually become well-known technology. Some advanced technology will have rapid promotion and communication. In this environment, enterprises should break through the single industry or field limit, study and introduce all kinds of new technology, and design a set of general automation system frame.

This article first proposed a general automation system design, and then analyzed several emerging technologies suitable for the automation system. At last, it analyzed the characteristics of this system and looked forward to the future technology development of the automation system.

The Design of The Automation System

The automation system in the future must have good ability of information acquisition and processing, the ability of the traditional information processing, the ability of time series information processing, the ability of data compression and access and better visualization.
technology of user experience. A set of frame structure will be shared, and the industries can choose different key technology according to their own needs. Picture 1 is the framework of the automation system.

Information Acquisition. There are three ways of information acquisition in general.

1) Network Communication. Through the TCP/UDP protocol, network communication can access the remote information according to some protocol. The mainstream standard protocols are usually recommended.

2) File Interaction. According to the current industry experience, file interaction with the self-explanatory ability is recommended, such as XML format, E format, etc.

3) Manual Entry. The man-machine interface must be convenient for user to use. It is recommended that the man-machine interface has the intelligent association function.

Information Processing. Information processing changes the fastest in the technologies of automation system. The information processing need to support different operating system, different database, and focus on the time-series data processing. The design of information processing needs to consider the following points.

1) Real-Time Requirements. Different real-time requirements have different design ideas. The information management of manual entry usually uses the commercial relational database directly, such as ORACLE, MYSQL, etc. And the high real-time demand system needs to consider using the real-time database based on memory management. Real time database can use more mature database such as FastDB, ExtremeDB, SQLitek, TimesTen, etc.

2) Massive Information Processing Requirements. Massive information processing is divided into two aspects. One is the read and write of the real-time database based on memory management. The traditional HASH algorithm can improve part of read and write performance, but in the specific operation, through the design can greatly improve efficiency. Tab.1 and Tab.2 are two design ideas of time series database. They can meet different efficiency requirements.
according to different applications. When you need to take a time cross section data of all POINT, as shown in Tab.1, you can copy the whole contiguous memory to achieve the highest efficiency. When you need to take a one minute time series data of the same POINT, as shown in Tab.2, the structure of Tab.2 is more conducive to obtain the data than Tab.1.

Table 1 Time-series of real time database of longitudinal sequence

<table>
<thead>
<tr>
<th>TIME</th>
<th>QUALITY</th>
<th>POINT0</th>
<th>POINT1</th>
<th>…</th>
<th>POINTn</th>
</tr>
</thead>
<tbody>
<tr>
<td>12:00:00.000</td>
<td>0</td>
<td>311.3</td>
<td>1311.3</td>
<td>50.3</td>
<td></td>
</tr>
<tr>
<td>12:00:00.020</td>
<td>0</td>
<td>313.5</td>
<td>1313.5</td>
<td>50.5</td>
<td></td>
</tr>
<tr>
<td>…</td>
<td>0</td>
<td>312.4</td>
<td>1312.4</td>
<td>50.4</td>
<td></td>
</tr>
<tr>
<td>12:00:59.980</td>
<td>0</td>
<td>316.3</td>
<td>1316.3</td>
<td>50.3</td>
<td></td>
</tr>
</tbody>
</table>

Table 2 Time-series of real time database of transverse sequence

<table>
<thead>
<tr>
<th>POINTID</th>
<th>QUALITY</th>
<th>12:00:00.000</th>
<th>12:00:00.020</th>
<th>…</th>
<th>12:00:59.980</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>0</td>
<td>311.3</td>
<td>313.5</td>
<td></td>
<td>316.3</td>
</tr>
<tr>
<td>1</td>
<td>0</td>
<td>1311.3</td>
<td>1313.5</td>
<td></td>
<td>1316.3</td>
</tr>
<tr>
<td>…</td>
<td>0</td>
<td>…</td>
<td>…</td>
<td></td>
<td>…</td>
</tr>
<tr>
<td>N</td>
<td>0</td>
<td>50.3</td>
<td>50.5</td>
<td></td>
<td>50.3</td>
</tr>
</tbody>
</table>

The other side of massive information processing is the storage and retrieval management of the historical data. When slow accumulation of information achieve the mass, the data should be storage scattered through the years, months, days, hours, etc. The single table over millions of records should be avoided. To the data of time-series of real time database in Tab.1 and Tab.2, the update frequency is up to 20ms, there will be 3000 section data every minute. So the information compression and fast storage should be considered and simple file database should be used [3].

3) High Speed Retrieval Requirements. In commercial relational database, the storage of massive information can improve retrieval performance through creating an index and storage processes. And in simple file database, the storage of information can achieve efficient information retrieval through B+ constant search algorithm to define and manage index. The technologies such as HADOOP can be tried to use in automation system.

Visualization. Visualization is the most direct way for users to understand the system; it is also one of the concerns of the user. In the practical application, there are complex applications using fat client technology of C/S architecture. There are also Web applications using thin client technology of B/S architecture. Requirements are increasingly high degree of identical. RIA (Rich Internet Applications) is born in such a background. It has highly interactive, rich user experience and powerful client. At present, WEB and desktop software field are gradually close to RIA. The most outstanding characteristic of RIA is “RICH”, and the core part of RIA is also in “RICH”. “RICH” contains two meanings:
1) **Rich Data Model**: A variety of data model is provided by RIA to handle the client complex data manipulation. Using RIA, it can transfer the background process problem to the client, so the data can be cached in the client, to achieve a user interface with faster response speed than that based on the HTML and less number of data to and from the server.

2) **Rich Interface Elements**: RIA provides richer interface elements than HTML. Intensive, fast response speed, rich graphic elements on the page and data modal together provide good experience for the user [5-6].

   RIA has the characteristics of desktop applications including: provides an interactive user interface in the message confirmation and formatting; provides quick response time under no page refresh; provides a common user interface feature such as drag-and-drop, online and offline operation ability. RIA has the characteristics of Web applications including: immediately deploy, cross-platform, using step by step download to retrieve the content and data and can make full use of the widely accepted standard of the Internet. RIA has the characteristics of communication including real-time interactive sound and image.

   The role of the client in RIA is not only a display page; it also can asynchronously calculate with the user behind the scenes, send and retrieve data, show the integration of the user interface and integrate use the sound and image. All this can be done not depend on the client connection to the server or the back end. At present, there are several RIA client development technologies such as Adobe Flex, Avalon, Java SWT, Bindows, JavaFX, SilverLight, etc. Among them, Adobe Flex support the most widely platform and application range.

**Message Service.** Application of message service technology makes it possible to exchange or integrate data that run on different machines. Various applications can exchange data through internal agreement or protocol. There are great differences of message services with different needs. For example, high-speed network message exchange through TCP or UDP has the characteristics of strong flexibility, high speed. But in most cases, there are many kinds of message which change rapidly, and the speed requirement is not high. So the support for different performance requirements message must be considered in the design process of message service. In many automation system, for security protection requirements the firewall equipment will be increased, which must be considered in message service design.

   The current popular Web Service, based on some of the conventional industry standards and the existing technology such as XML and HTTP, can reduce the cost of application interface; can provide a general mechanism for the integration of business process of the entire enterprise or among multiple organizations. In the design of automation system, it can provide the special exchange channel of TCP or UDP for the special needs; it also can select Web Service to provide message services to construct a set of general and large integrated system across the firewall.

**Key Techniques in the Field of Automation**

**The Time-Series of Real Time Database Technology.** The real time information management of automation system always uses the shared memory of real time database technology. But the traditional memory database only supports two dimensional managements of the simple row
combination. That is, when the real time information updates, it can only get the data of a time section just like taking photos, which is almost useless to the millisecond information update [3]. With the development of IT technology, millisecond level requirements are more and more, the information management of automation system require dynamic movie, not photos. So the time-series of real time database technology will be the essential technology of automation system.

The focus of the time-series of real time database technology is the memory management. It will use the memory with rolling circulation. One of the ways is shown in Table 1.

Tab.1 looks like the traditional two-dimensional table which makes no difference, but in fact information of each row in the table will be changed along with the time update. As in Tab.1 the information will scroll update for 12:00:01, 12:00:02 information, so the system can access to a time-dependent section sequence, not only a section. The example of Tab.1 can provide time-series data of one minute.

**Lossless Data Compression and Storage Technology.** The development of technology brings mass of information, and in automation field, for fault analysis, decision support and technical research, etc. it is used to retain a considerable period of time information. Relational database is a common way of data storage. With the help of high capacity storage devices relational database can do a lot of data storage; the massive data gradually challenge the relational database performance and the capacity of a storage device. Enterprises can't unlimited increase the hardware; they also are reluctant to give up the data. In addition to backup of database, lossless data compression technique is commonly accepted way in many fields.

The lossless compression algorithms, such as, Huffman code, run-length coding, LZ series dictionary method, etc. are commonly used. We can select the algorithm according to the different information characteristics of various industries. The key point of algorithm selection is that the performance of the compression and decompression meet the specific application requirements [2].

After the compression of massive data, there will be data block. The storage of data block can use BLOB, which is the relational database, or Berkeley DB, which is document database with the rapid retrieval capabilities, or memory database with file management, rapid retrieval ability, which is independent development of the B+ search technology. In order to meet the complicated retrieval conditions, it is better if the storage of data can support structure index.

**Visualization Technology Based on FLEX.** Flex usually refers to the Adobe Flex, which was originally released by Macromedia Company in 2004 March. Based on its proprietary Macromedia Flash platform, it covers a series of technical development and deployment that support RIA(Rich Internet Applications).

Flex uses the GUI interface; it uses MXML language based on XML. With a variety of components, FLEX can realize the Web Services, a remote object, drag and drop, sorting,
graphics and other functions. FLEX has animation and other simple interactive interface. With respect to the application based on HTML (such as PHP, ASP, JSP, ColdFusion, CFMX, etc.), in FLEX the client need to load the only once, the workflow of FLEX applications are greatly improved [4].

As a leader in the rich client technology a new generation of Internet, the Flex technology has been more and more used by the company, and more and more users and programmers accept it. It has the first-class interface technology, which makes it possible for the programmers to make FLASH. The support of bottom call and support of different operating system platform make the system integration simplified greatly. It gives users a high user experience value.

Analysis and Prospect
In this paper, several typical new technologies were discussed in the automation industry. It proposed a general automation industry automation system design scheme. The scheme represented the development trend of automation system, and outlined some design ideas of automation system of some industries. After further refinement, it can provide guidance to the development and design of the automation system.

References


THE ANALYSIS OF INFORMATION MANAGEMENT ON RENOVATION IN UNIVERSITY CAMPUS

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Abstract. Renovation in university campus is a complicated repair work, the face to the country allocated funds, the size of the school project and other projects, the orderly implementation of the project funds, how to control the overall implementation of the project better is an important and detailed work. Based on the analysis of the restoration process of the school, it is proposed the establishment of a university repair management in line with the logic of information management platform in this paper, to a reasonable solution to the management process involved in the restoration to the people and affairs.

Keywords: component, Information Technology, Repair Management, Standardization Process, Scientific Decision.

Introduction

With the development of the reform of China's universities, many colleges and universities to expand multiplied the campus, at the same time, the old campus and new campus in warm water, electric and other infrastructure during the conflict, highlighting the overall planning, which requires appropriate and reasonable repair transformation. Do a good job on restoration and improvement of work processes, information-based workflow to achieve, to do good repair management, effective control of repair progress and cash flow, at the same time to improve work efficiency of the significance.

The Status of University Logistics

A. The building of repair management information processes, need renovation of all aspects of the analysis, in order to establish and improve the repair management system, the establishment of information public, effectively avoid risks, to ensure the quality of construction, improve the efficiency of capital use. First analysis of the management of common repair status of colleges and universities:

Lack of rules: The prevalence of most colleges and universities renovation management system is imperfect, the low level of information disclosure and practices. Some colleges and universities have developed the restoration and improvement project management approach, basically from the frame to limit, but the lack of support in the implementation process in details. Settlement after the end of the project and the audit process, the audit department in the works could not understand the process, resulting in the audit department in a subordinate position.
Therefore, the repair project management needs from the project, bidding, contracts, construction management, settlement and auditing process of the development of relevant rules.

B. Periodically: Public housing for the university resources and facilities and other aspects of the restoration and improvement, the loss due to natural causes, need to periodically for the water and electricity, gas, warm, doors, windows, interior and exterior decoration.

C. Complexity: The restoration and improvement involves more extensive. On one hand is related to the department, staff are more related to the approval and implementation procedures is relatively complicated, in addition to basic infrastructure department is responsible for building schools, but some schools there are logistics group projects, logistics and asset management projects, and Education Department of special projects funded, the college also has its own repair and renovation funds, etc.; on the other hand, the type of complex renovation project, more related to technical expertise; third is the construction involving a wide range of repair, the construction of the organization planned is random in particular.

D. Contradiction: As the school's enrollment, infrastructure investment and management of repair is inversely proportional to the input of funds, in addition labor costs and material costs rise, exacerbating the difficulty of the restoration and improvement during recent years, it is important to consider how to use the limited restoration funds to provide efficient repair management services into new contradictions. Part of the renovation project by a number of sector management, lack of interdepartmental communication, likely to cause conflict between departments.

E. Unexpected: To ensure the school's teaching, research and student life in normal operation, on one hand the need for timely handling emergency situations, such as: housing leakage, damage to the glass curtain wall, etc.; on the other hand, the relative concentration of the time the restoration and improvement general on the implementation of summer vacation, the other school sponsored activities, will increase the temporary restoration and improvement.

F. Lack of planning: Although the planned renovation project at the end of collection, but does not guarantee that the project be implemented within the planned, unplanned items are not construction. Repair and management in colleges and universities, only large-scale restoration and improvement project was carried out pre-consultation discussions, determine the project's feasibility and implementation. In addition, there is a period of time, repeated repair work situation, lack of planning in the management of the process.

**Repairing Information Technology Management**

Dealing with the repair problems of management process, and combined with the experience in the repair management in NCEPU, this paper presents a restoration management of information technology management concept, the specific implementation of the framework are as follows:
A. Improve the Repair Management System, Regulate the Audit of the Regulatory. This helps maintain the concentration of repair management, unity, maintaining the relevant departments (Logistics and Asset Management Department) the authority to overcome the complexity of the work process and randomness, improve quality and efficiency. Management system throughout the management process to refine the various stages of analysis from the demonstration projects, feasibility reports, leadership approval, project approval, site reconnaissance, identification of quantities, bidding, contracting, construction, change negotiations, covert project acceptance, final acceptance, project settlement, disbursement of funds, project quality and other aspects, to develop detailed regulations and regular inspection of the specification.

Meanwhile, the department should establish audit tracking system, tracking monitoring is reflected in pre-project, mid-term and later. In the pre-site visit, to use the unit with the audit departments to determine quantities and construction drawings, etc.; the tender and evaluation, there are audit staff to monitor and discipline inspection departments involved, a reasonable proposal, play a multiplier effect; to negotiate the contract and project construction and subtle process, should be invited to participate in the audit department of the site management, this audit can grasp the scene first-hand information on the validation control to play the role of project cost; after the project acceptance, need to provide engineering construction unit the accounts, together with the construction process of the relevant information to the audit department, the project management summary of the test; finally project concluded in the audit department issued a single case, according to the contract for the payment of project funds. As the renovation involves the allocation of funds, so the project audit process, the relevant person in charge of economic responsibility into the scope of the audit, examining the level of project management, honesty and self-discipline and use of funds effects.

B. Unified Reporting, Unified Planning, Unified Implementation. Management of a system of linkage repair work requires mutual cooperation, the project itself is a cyclical process for this phenomenon, the university should identify a lead department, in complying with repair management system developed under the premise of the establishment of information management platform, to form a linkage mechanism to work. In order to fund the main line, in the approval and monitoring processes to increase control authority, to achieve unified management. As the school's department responsible for managing the repair is within the mandate of the school, responsible for implementation and restoration-related work, the management of the repair work as the department's main content in the school's overall logistics system constraints, adhere to the leadership responsibility system, the annual repair works as a department work plans, periodic research, discussion, supervision, set up a dedicated repair management posts, the duties of this job is to protect the information platform for the main line work, the project's daily affairs, the exchange of coordination, repair process tracking and management, learning professional knowledge, understanding, practice and technology play a regulatory role in strengthening the management of the construction unit and guidance in project formulation and approval, preparation information, the contract was signed, construction management, organizational acceptance, settlement and approved the submission of audit
information, etc. play an advisory role with the audit department at work, engineering and construction unit of the amount determined, the approved settlement coordinating role.

C. Process Management. Based on the actual renovation of management concepts, this reference to the university’s specific workflow, renovation work will refine the process, according to planned, unplanned preliminary budget for the renovation project, and then follow the appropriate rules and regulations for repair management. Specific workflow is shown in Figure 1.

D. Set Up Repair Management of Information Platform. Section mentioned above flow chart, based on the establishment of information management platform, mainly due to the information requirements and file summary information, the purpose of information disclosure, supervision.

With the development of information technology, communication between departments in order to overcome the difficulties are not smooth, to build a unified information platform to help repair the management of information resource sharing, to help supervise the project and file archiving information and inquiries, timely exchange of information and deal with emergency cases, the use of information access control platform coordination between various departments. As the lead department for repair management (logistics and asset management office), the most basic job is to collate information on file with the same common management, centralized management of engineering information can save a lot of independent duplication of the information in this platform construction project, requires that all paper records of electronic information, according to project type, nature, settlement, etc., will focus on electronic records stored in the information platform of data centers to achieve a restoration project microcosm of the entire process paperless, When you need to check a project, from planning to completion of the process and the resulting data will be displayed in the corresponding man-machine interface, thus improving work efficiency. In each user login and operation of the process, through the user access control, users of information at all levels of the submission process has a full record, its value lies in communication between different departments to provide a bridge, and to provide for the management of late reference. As the leading sector of the school’s infrastructure, it must to fully grasp the situation, housing, etc., so that electronic information is updated dynamically, for the future of housing management, maintenance and other work-based program development and implementation basis.

Repair work in accordance with the management process to project the amount of budgetary control as the basis to determine the bidding situation to the school, for example, specific provisions shown in Figure 2.
Each department reports the Forms of repair plans

Approval by directors of school leaders

Logistics and Asset Management Office setting up project

Preparations for bidding

Projects bidding

Agreement signed

Construction site management

Project acceptance

Auditing for project settlement

Annual repair notice issued

Unplanned repair project

Fig. 1 Repair management flow chart.
E. Improve the Professional Quality and Technical Level of Manager. In the modern management concept, management to improve efficiency and create value in the management of repair involves many highly professional work, such as: project budget, technical jobs diversification, which management theory and practice requirements are high, the other repair management to update their knowledge quickly, and with the market fluctuations and changes in working conditions will also affect the case management strategy, thus requiring the management of workers engaged in repair and always maintain the state of learning, raise their political consciousness of the vigilance. The normal operation of the school renovation management, also need to strive for leadership at all levels of attention, set in the job responsibilities, personnel and other basic organizational structure to strengthen management, maintain a positive staff attitude and innovative thinking all the time. Reflected in the restoration management information platform construction, is to ask the Internet technology, real-time control of labor, materials, equipment and other market conditions, able to skillfully operate a budget estimate software project cost estimation, learning basic principles of operations research and management in the actual construction be applied in the comprehensive management of the affairs and progress of the project.
Conclusions

As a security-based teaching, scientific research, protect the normal operation of school infrastructure, the renovation project, not only fragmented complex, but also the amount of money involved are large, involve many departments, so the general idea of the project should be based on school development the overall situation, and orderly work effectively, and establish the concept of service and efficiency ideas. Learning the rules and regulations to strengthen and improve professionalism, increase the level of scientific management, good schools, clean and efficient basis for renovation.

References


A FAULT DETECTION METHOD FOR COMBINATIONAL CIRCUITS

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Abstract. As transistors become increasingly smaller and faster and noise margins become tighter, circuits and chip specially microprocessors tend to become more vulnerable to permanent and transient hardware faults. Most microprocessor designers focus on protecting memory elements among other parts of microprocessors against hardware faults through adding redundant error-correcting bits such as parity bits. However, the rate of soft errors in combinational parts of microprocessors is considered as important as in sequential parts such as memory elements nowadays. The reason is that advances in scaling technology have led to reduced electrical masking. This paper proposes and evaluates a logic level fault-tolerant method based on parity for designing combinational circuits. Experimental results on a full adder circuit show that the proposed method makes the circuit fault-tolerant with less overhead in comparison with traditional methods. It will also be demonstrated that our proposed method enables the traditional TMR method to detect multiple faults in addition to single fault masking.

Keywords: Soft Error, Transient Fault, Fault-Tolerance, Combinational Circuits, Full Adder.

Introduction

As the transistor dimensions have shrunk and the large-scale integration in electronic switches has increased, chip fabricators can insert more than one billion transistors in a single chip. Such integration scale can increase the performance of chips. Many of new architecture techniques, such as Superscalar and Chip-Multi Processor (CMP), actually need such number of transistors. However, the ever-increasing nonlinear power consumption in the technology trend could be a disaster for circuits, because the transistor density is going up intensely. To prevent this problem we should decrease the voltage supply, and this change can lead to falling the noise margin in the circuit [1].

On the other hand, by shrinking the feature size, the factor QCritical (electrical charge of capacitances) is decreased too, and this problem can lead to increase the probability of fault occurrence in the circuit [2]. It is proved that large-scale circuit integration increases the failure rate exponentially [3]. Generally, in new generation technologies, we have less reliability than the old ones. Some of the reasons for these problem are: lower CL (load capacitance), lower VDD
or VCC (supply voltage) that lead to a smaller noise margin, lower QCritical, more process variation [4] and manufacturing defects[5].

These factors affect the reliability, that is a key concept along with performance and power metrics, and needs to use a fault-tolerance mechanism [3, 6, 7]. Typically, all components of chip can be classified in to two categories, Logic Block sand Memory Elements.

Commercial microprocessors typically use Error Correction Codes (ECCs) to protect these circuit elements. ECCs, such as parity, add latency to each access and results in an appreciable performance penalty, moreover, it is difficult to implement for logic blocks [8,9]. However, combinational circuits are very importance for fault-tolerant design. Because new technologies are facing less degree of electrical masking and this phenomenon makes circuits more susceptible to faults [10]. In [3] it has been mentioned that from the year 2011 on importance of improving fault coverage in combinational circuits will overcome the sequential ones. For this reason, we decided to choose this area for the implementation of our technique.

Fault-tolerance techniques are generally accomplished by using redundancy in hardware, software, time and information [11]. In this paper, we have used hardware redundancy in combinational circuits. Some of the fault-tolerant hardware methods are Duplication With Comparison (DWC) in which the module is duplicated, result are compared and if one mismatch occurs, an error flag is raised. N-Modular Redundancy (NMR) design techniques add reliability to a system at the expense of extra hardware resources. In an NMR system, all protected modules must be replicated N times, in order to allow for automatic masking of N/2 of faults happening in separate modules. Standard Triple-Module Redundancy (TMR) methods are used frequently. Using these methods, triple modules and voting circuits are implemented onto an Application Specific Integrated Circuit (ASIC) or a Field-Programmable Gate Array (FPGA). When a fault occurs, the voting circuit neglects the value of a faulty module and takes a correct value of the other two non-faulty modules. These methods come with high area and power dissipation penalties and are inherently proposed for detecting or masking a single fault [11]. This paper is organized as follows: section 2 presents a brief background of fault sensitivity in combinational and sequential circuits; in section 3 we proposed a new fault-tolerance technique in combinational circuits; in section 4 we apply this method to full adder circuit; and finally, in section 5 concluded.

**Background**

Single event transient pulse is induced when cosmic particles such as Neutron Strikes, or radiation from packaging materials such as alpha Particle with enough energy hit the sensitive region in the circuit [10]. The voltage pulse propagates through an activated path in the logic circuit. When it is captured by a clock edge, a soft error occurs. Otherwise, that pulse is called a transient fault [12]. In recent years, with advance in the technology of fabrication, transistor quantities, processors are becoming increasingly vulnerable to transient faults [13]. Transient faults currently account for over 80% of faults in processor-based devices [14]. In a typical
integrated circuit, memory arrays, latch elements, and combinational logic are the most sensitive parts and could be affected by soft errors and transient faults.

Historically, soft errors were a concern in the design of memory elements, but the susceptibility of the combinational blocks to transient faults increases as a side effect of technological scaling. Combinational logics are usually used for designing arithmetic circuits (such as adders, multipliers, etc) or in other words the data path of a computer.

We should know the importance of employing combinational circuits in applicable chips processing is rising, as they are simpler, operate faster, and consume less power than sequential ones. Moreover, many of statistical researches proved this statement [15, 16]. Combinational circuits occupy a considerable portion of processing chips in comparison with sequential circuits. For example in FPGAs, The ratio of using combinational circuits to sequential ones varies between 5 to 100 times [17,18].

Continuous device scaling, higher degree of pipelining and decreasing electrical masking effect, contribute to the increase in soft error rates in combinational circuits [10]. Transient faults in combinational circuits are catching up with errors in memory elements [3]. A transient fault in a logic circuit might not be captured in a memory circuit, because it could be masked by one of the following three phenomena [3,19,20]: First, Logical Masking, occurs when a particle strikes a portion of the combinational logic that is blocked from affecting the output due to a subsequent gate whose result is completely determined by other input values. Second, Electrical Masking, occurs when the pulse resulting from a particle strike is attenuated by subsequent logic gates due to the electrical properties of the gates to the point that it does not affect the result of the circuit. Third, Latching-Window Masking, occurs when the pulse resulting from a particle strike reaches a latch, but not at the clock transition where the latch captures its input value.

These masking effects have been found to result in a significantly lower rate of soft errors in combinational logic compared to storage circuits in equivalent device technology [3]. However, these effects could diminish significantly as feature sizes decrease and the number of stages in the processor pipeline increases. Electrical masking could be reduced by device scaling because smaller transistors are faster and hence may have less attenuation effect on a pulse. Also, deeper processor pipelines allow higher clock rates, meaning the latches in the processor will cycle more frequently, which may reduce latching-window masking. Hence, in this work we focus on occurrence of transient faults and soft errors in combinational logic circuits and suggest a logic level fault-tolerant design method.

**New Approach Framework**

In this paper we presented a new approach to design fault-tolerant combinational circuits. Assume a logic circuit with m-input and n-output lines. Each output is a logic function of inputs. In this method, we use hardware redundancy to add a redundant output signal to the circuit. This new output generates the parity bit for output set. The value of redundant output is directly
derived from the input lines. There are two main types of parity checking in digital systems, Odd Parity (Po) and Even Parity (Pe). Both of these types can be used in our scheme. Because parity checking mechanism is a relative method and it is sufficient that both sides be aware of the convention of data communication [21]. If we model the input/output sides of a logic circuit by the sender/transmitter stations in a telecommunication system, we can say that parity checking is a conventional method to check the bit errors in the telecommunication systems. In this technique, the transmitter station tries to send an extra bit accompanied by transmission data bits, in order to detect single bit errors that may occur on the channel after getting data by the receiver station.

In our scheme, first we calculated the truth table of the redundant output line as an even/odd parity for output lines. Then, try to make it related to the input bit arrangements. After finding the function role and simplifying it, we can plot by the least needed logic gates. It is important that we should not use the middle terms of the main circuit to design this redundant line. If middle term are used, faults occurring before these branches would not appear on the output (i.e. the redundant parity signal). If we assume using the even parity mechanism for designing the redundant output line, XORing this line with the other main circuit outputs can demonstrate the error occurrence. We name the result of this XOR gate as the ERROR signal. Seeing zero in this line means that no error has occurred, and seeing one can refer to an error in a part of the circuit. It is clear that by using the odd parity checking mechanism XOR gate converts to XNOR gate, but the outline of overall method remains as before. The framework of this scheme has been shown in Fig. 1.

TMR method is a conventional technique to design fault-tolerant circuits. This technique can mask a single fault by voting results. But, if multiple faults occur in the modules, this method is unable to mask or detect them. As a result, incorrect outputs are voted and an error is appeared as the result of the circuit. Hence, TMR treats weakly when facing multiple faults.

Replacing our proposal module with conventional TMR modules can help to detect multiple faults by voting ERROR signals in addition to the other output lines. It worth emphasize that our proposed approach is capable to detect all single faults that may occur in the logic circuit, whether they lie in the main part of circuit or not.

![Fig. 1 Overall new approach framework.](image_url)
A Case Study

ALUs are very important combinational circuit block that lie on the computational data path in the processors. The reason for their importance can be related being on the critical path. Typically, a key point of delay propagation in the processors is the maximum length of path between source registers and destination ones[22]. This length is depended upon the number of ALU functional units (FUs) that are lied on this computational path. Critical path on the processors should be designed as a fault tolerance path in order to increase reliability.

A. Designing Fault-Detection Full Adder (FDFA). To show the effects of our approach in practice, we used a FDFA an applicable logic circuit in the ALUs. FA is a simple logic circuit that is used as a basic element to design many of functional units, such as adders, subtracters, multipliers, dividers, etc. All circuits in our experiments are custom designed and laid out in 1.8V, 0.5µmCMOS technology and simulated by using HSPICE tool. This circuit uses a redundant part to provide an even parity for outputs SUM and COUT, which is named E. E is a logic function of FA input lines, named A, B and CIN. The value of this output line is selected in a way that the number of 1 bits in three output lines, always be even. Table 1 shows the truth table of FDFA.

![Gate-level FDFA](image-url)
Table 1 Truth table of FDFA

<table>
<thead>
<tr>
<th>A</th>
<th>B</th>
<th>CIN</th>
<th>SUM</th>
<th>COUT</th>
<th>E</th>
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</thead>
<tbody>
<tr>
<td>0</td>
<td>0</td>
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<td>0</td>
</tr>
</tbody>
</table>

\[ E = A \cdot B \cdot CIN + \overline{A} \cdot \overline{B} \cdot \overline{CIN} = \overline{A \cdot B \cdot CIN} \cdot \overline{A \cdot B \cdot CIN} \] (1)

After deriving the value of E line for all input arranges, the equation (1) simply is resulted. This equation shows that in order to implement E line, three NANDs and three NOTs are totally needed.

In the next stage, we try to design this circuit by above gates. The derived gate-level design of FDFA has shown in Fig. 2. After it, we sketch the layout of this circuit in the L-Edit tool and get the waveforms of FDFA operation with giving some experimental inputs. Fig. 3 shows these waveforms for all of input and output lines.

B. Implementation Results. We demonstrated the hardware and timing overheads of using FDFA in comparison with the NFA. Next, with adding a XOR gate into the FDFA (as XFDFA), the calculated overheads compared with the DWC method with NFA modules.

Table 2 shows area, propagation delay and dynamic power consumption overheads. The area overhead is about 27% that is the result of 12 additional transistors in FDFA design. About propagation delay, that is a very important factor to evaluate circuit specifications, both of schemes are similar. Because worst-case delay is limited by path of generating SUM signal. In TABLE III, we compared XFDFA with DWC Figure 2. Gate-level FDFA because both of them are capable to detect a single fault.

In order to implement DWC, we use two coupled NFAs with XORed similar outputs together to check results. Our XFDFA is designed with 28 transistors less than DWC and almost can save 28% in dynamic power consumption. Because the redundant component in DWC is greater than XFDFA. But, in delay we are penalized about 37%. The reason of this penalty is using a two level XOR in the final stage of XFDFA. If we verify the arrangement of gates in each circuit by using logic level viewing, find out that both of these methods use a 2-level combinational circuit in their output side, but the last gate in DWC (an OR gate) is quicker than the end gate of XFDFA (a XOR gate).
Table 2 NFA vs. FDFA

<table>
<thead>
<tr>
<th>Metric</th>
<th>Area</th>
<th># of Transistors</th>
<th>Delay</th>
<th>Power (fwT)</th>
</tr>
</thead>
<tbody>
<tr>
<td>NFA</td>
<td>204*58 λ²</td>
<td>38</td>
<td>54τ</td>
<td>2.68</td>
</tr>
<tr>
<td>FDFA</td>
<td>254*58 λ²</td>
<td>50</td>
<td>54τ</td>
<td>3.86</td>
</tr>
<tr>
<td>Overhead</td>
<td>24%</td>
<td>31%</td>
<td>0%</td>
<td>44%</td>
</tr>
</tbody>
</table>

Table 3 DWC vs. XFDFA

<table>
<thead>
<tr>
<th>Metric</th>
<th>Area</th>
<th># of Transistors</th>
<th>Delay</th>
<th>Power (fwT)</th>
</tr>
</thead>
<tbody>
<tr>
<td>DWC</td>
<td>339*215 λ²</td>
<td>92</td>
<td>131.2τ</td>
<td>6.387</td>
</tr>
<tr>
<td>XFDFA</td>
<td>311*128 λ²</td>
<td>64</td>
<td>179.8τ</td>
<td>4.981</td>
</tr>
<tr>
<td>Improvement</td>
<td>45%</td>
<td>30%</td>
<td>-27%</td>
<td>22%</td>
</tr>
</tbody>
</table>
Conclusions

In this paper, we proposed a new approach to design fault-tolerant combinational circuits. The main idea behind our proposed approach is using a redundant circuit that operates as an even parity generator for main output lines. We also showed that traditional methods such as TMR detect multiple faults in addition to single faults if they are combined with the proposed method. Experimental results obtained from testing the proposed approach on a full adder circuit exhibit about 27.5% overhead in area and 44% penalty in power consumption, which shows about 37.5% improvement in area and 22% in power over the traditional DWC method. This method shows to perform almost equal to DWC from propagation delay point of view.

References


AN ADAPTIVE TUNABLE RAMAN SPECTRUM INTENSITY METHOD BASED ON DIGITAL AND ANALOG AGC CONTROL

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Abstract. This paper presents a auto control Raman spectrum intensity method based on auto adjust linear CCD integration time and analog amplifier gain control. Introduces a linear CCD timing sequence control signal generation and integration time auto control realization. Also present the linear CCD analog signal output auto gain control circuit and realization. Publishes this design key parts analog and digital circuit schematic, provides a most effective solution for hand held Raman spectrometer fast measurement and lower power consumption.

Keywords: Linear CCD, Integration Time, Digital and Analog AGC control.

Introduction

Hand held Raman spectrometer is being widely used nowadays and Raman spectrum signal intensity directly affects optical spectrum analysis performance. Raman spectrum signal intensity is directly account on three facts: 1. linear CCD optical pixl exposure time, 2. laser source power, 3. optical constructure. Optical constructure can not dynamic adjust. It is fixed after design. So usually two kind of way could be often used to change the Raman spectrum signal intensity: increase the laser source power, or increase CCD integration time. But increasing laser power is limited by laser maximum output power. Also over high power laser power will burn the sampling products. So increasing the CCD integration time is often simple way. But, this kind of method have to increase instrument measurement time and more laser power consumption, also reduce laser life time. So this paper presents a better soulation: combination the integration time auto control and analog amplifier gain auto control.

Whole system level hardware design digram is showed as figure 1. In this design, ILX511 Linear CCD is main control target, in order to obtain high SNR and high resolution in this system, a high performance 18 bit SAR ADC chip AD7641 is used, and in order to realize the analog auto gain control, a digital potentiometer is used, and a serial port to parallel conveter mode also is used in FPGA, an asynchronized FIFO RAM has been used as bridge between the ADC and DSP, FPGA is used to generate variety CCD timing clock and asynchronized FIFO control signal. TMS320VC5509A DSP is used as a System Design Based on FPGA and DSP whole system control unit, because of it’s fast data processing ability, low power consumption and built in USB 2.0 port. In the system, DSP send a initial integration parameter and start optical spectrum data sample, after DSP obtaines the sample data though ADC conveter, use a feedback algorithm to calculate a average signal intensity, and then using this average signal intensity to compare to prepared intensity setting value, and using this compared difference as a feedback error, then calculate a new integration time setting value and send to control analog amplifier gain and FPGA CCD timing.
**Linear CCD Time Sequence Signal Generation and Optical Spectrum Data Acquisition**

**A. ILX511 Linear CCD Time Sequence Signal Generation.** The ILX511 is CCD linear image sensor, its feature: 2048 pixels, 14 µm x 200 µm (14 µm pitch), Single 5 V power supply, ultra-high sensitivity, built-in sample-and-hold circuit, maximum clock frequency is 2MHz, its wavelength range from 400~1000nm, it is well fit to Raman Spectrum data analyzing wavelength range, when use a 785 nm laser source.

Figure 2 shows this CCD detector time sequence diagram, figure 3 shows its analog output waveform, so in order to drive ILX 511 CCD, external driver circuit has to provide two time sequence control signal: one is CCD clock signal ΦCLK, it's maximum frequency is 2Mhz, typical working in 1Mhz. another is timing driver signal ΦROG. The ΦROG pulse period decides the scan a frame pixel time and CCD exposure time, since a frame has 2048 pixel + 38 dummy signal =
2086 clock pulse, so scanning a frame CCD pixel needs at least 2086 clocks signal period, and the true effective clock generated analog signal, which is sensitive optical exposure source, has only 2048 pulse. so CCD time sequence signal generation module has to have function as follows: 1) gernerates ΦROG pulse, ΦROG period > 20486 X ΦCLK period; 2) generates more than 2086 clock pulse ΦCLK; 3) generates 2048 clock pulse with duty cycle 50% ΦADC_CONST, which is corropsed CCD output analog signal and synchronized to start ADC convter. The CCD_ROG negative pulse enable the CCD output effective and let the CCD exposed pixl convter analog signal output by CCD clk clock driving. So how to create these three control pulse is the design key point.

Figure 4 displays this three control pulse waveform, and figure 5 presents their FPGA loglic function schematic. In this design, a stable 20Mhz clock source can be obtained though FPGA internal PLL. Divide this 20 Mhz clock source into 1 MHz with 50% duty cycle using 10 divider, this 1Mhz clock function as whole CCD driving clock source, the CCD and ADC time sequence control signal generation module is show in Figure 5. In order to obtain a ΦROG signal, use a 4 divider to get a 8 us width pulse as a ROG generator, the ΦROG period is controlled by reciving parameter though McBSP port coming from DSP.

In Figure 5, module inst20 implement the ΦROG pulse generation function, module inst10 implement the CCD clk and ADC start pulse signal. The figure 6 shows this signal generation simulation waveforms. As figure 6 show simulation result, the function is exactly well meet the CCD control time sequence reguirment.

![Fig. 4 CCD Control Signal Generation Diagram.](image)

![Fig. 5 CCD clk, rog and ADC const control pulse generation FPGA schematic.](image)

![Fig. 6 3 main CCD control pulse waveform simulation result.](image)
B. Programable Integeration Time Control Mode Realization. From figure 7, CCD analog output voltage is proportional it’s integration time, when integration increases the analog output voltage also increase, but how to change the CCD integration time and keep the ccd clk is the same frequency is the design key point. In this design, a programable counter is used to implement the CCD integration time adjustable. As figure 7. schematic show, divide the input CCD clk and obtain a 1khz pulse, and then use D latch the input with the input clk_in1, which control the clk rog duty cycle, and use and gate get a rog pulse. So input counter DVF decides the clk ROG period. that mean, change the counter value, can change the ROG pulse period, the period is determined by clk divider, in this design, it is set in 1ms, so integration time step unit is 1ms, for example, if want to set integration time is 10 ms, only needs send data 10 into counter, the programable integration time generation mode FPGA schematic is presented in figure 6. In figure inst3 receives integration time set value though SPI convter to parallel function mode, and send a 16 bit width data bus to programable counter inst9, and then obtain tunable CCD ROG pulse.

![Output voltage rate vs. Integration time (Typ.)](image-url)

Fig. 7 CCD Integration Vs output voltage rate.

![Programable integration time generation mode](image-url)

Fig. 8 Programable integration time generation mode.
Programmable counter simulation waveform.

Programmable integration Time generation simulation waveform.

C. CCD Analog Output Autogain Control. Figure 11 shows the linear CCD analog voltage signal output amplifier circuit, the analog signal go though a follow amplifier, and differential OPA converters, get a positive signal, and then use a non-convert OPA to amplifier the sample voltage signal. the gain is determined by R5. Rp, when factually use, R3 could be discarded, Gain = 1+Rp/R5 . Rp is the potentiometer resistor value, which is set by DSP I2C bus. The ADC7641 ADC chip is an 18bit high performance SAR ADC with SNR 93 db, 2 MSPS, and its power supply only has 2.5 V and differential input range is ±2.5V, also it’s power consumption has only 75mw, this extraordinary features is especially well suitable for portable and handheld Raman optical analysis instrument. In this design set ADC mode in 16 bit parallel, and sampling mode set in normal 1.5 MSPS.

In order to implement the CCD analog voltage amplifier gain control, a 128 position potentiometer is used in this design. It use I2C bus communication with DSP, from above B intruded, this Raman spectromter integration time step unit is 1ms, so change the gain control speed do not need a fast communication, using 400kzh I2C bus speed is suitable for integration time change.
Conclusions

Utilizing this combination method, Raman spectrometer measurement time can be auto work in a best time range, so it can reduce 1/3 time to complete a sample test, and can save almost 1/3 battery power for laser drive, this method provides very valuable solution and it’s being successfully used in our new products.

Fig. 11 CCD analog gain control schematic.

Integration Time and Analog Gain Control Firmware Flow Chart.

Fig. 12 DSP firmware flow chart


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APPLICATION OF MATHEMATICAL THINKING MATERIALS TECHNOLOGY

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Abstract. Mathematical thinking is the essence of mathematics teaching, mathematical thinking in the basic concerns of modern mathematical content of the penetration and integration, enabling students to deepen their understanding of basic knowledge, understanding and the nature of mathematical developments, will help develop the students thinking tension, and expand their horizons; students for further study in mathematics, mathematical thinking and problem-solving view basis. This article discusses the functions, collections, dialectical, and modern mathematical modeling of the four basic mathematical thinking and content integration points.

Keywords: Modeling, Modern Mathematical Ideas, Basic Mathematics, Integration.

Introduction

In basic mathematics proper penetration of some modern mathematical thinking, mathematical thinking of students improve their math skills and literacy of positive significance; analysis to explore the mathematical basis of modern mathematical thinking and content integration point for teachers to a higher point of view look at the content and organization of teaching and the students understand and grasp the essence from the mathematical to provide effective help, which can improve the effectiveness of classroom teaching.

The Function of Ideas in Modern Mathematics Textbook in the Penetration of Basic Mathematics

Ideas of modern mathematics and basic mathematical functions at the integration of more reflected in the most basic math and algebra part number, a concrete manifestation of the function of the corresponding rules.

For example, in the basic content in the first grade to third grade, the number of students in learning by understanding the way physical and digital mapping to enable students to recognize numbers, when used in computing basic segments corresponding to illustrate and explain the minus sign to the students and the subtraction represent meaning. This kind of approach one can use a more intuitive way to help students learn, on the other hand reflects the thinking of modern mathematical function of the principle of equivalence. Through practice, after the students understand the function is helpful. For those students in lower grades, because the number and the number in the understanding of addition and subtraction, the intuitive, at this stage, the corresponding function of modern mathematics students thought likely to be recognized.
Secondly, after the third grade students also have the statistical knowledge of the function of thought in them. In the basic materials, such as bar charts, bar charts by observing that, students can clearly identify the number of men and women in various workshops; their understanding of the process contains a function on the corresponding ideas. Line charts despite a change in form, but the basic understanding of the content and methods, or bar charts and line, there is a corresponding thought in which, and line charts is to study the function of the image after the approximate figure. Students in the learning process, both to learn statistical methods, while there has been this kind of thinking corresponds to a better understanding.

Third, the basic mathematical knowledge in geometry and graphics, the graphics of the perimeter and area calculation process, also includes a function in which the idea. For example, in the basic fourth-grade rectangular and square area of study, when the formula, including an exercises is about the relationship between area and perimeter, in the same area of the case to the square of the perimeter of the shortest. Studied for a quadratic function of the high school students, through the analysis of the parabola to find the minimum, it is easy to solve. However, the lower for students who are not easy to understand, only through the media support, changes with software rendering, so that students understand the intuitive easy to accept. This reflects the function of the dependent variable and the relationship between the independent variables. In this form, students can learn to understand the basis for this method to the analysis of knowledge and reasoning, greatly enhancing their self-exploratory learning.

A Collection of Ideas in Modern Mathematics Textbooks in the Integration of Basic Mathematics

Collection of ideas is important and fundamental of modern mathematical thinking. Its concepts and methods have penetrated into the various branches of mathematics and other natural science disciplines, the development of these disciplines to provide a ground-breaking approach. Collection of basic mathematics content and the integration of thinking are everywhere visible. In the basic understanding of the number of first grade when the material is generally arranged for some, such as apples, small animals and other objects, to enable students to count in-kind by the combined number of elements to achieve the purpose of understanding. As students new to mathematics, their understanding is still very vague, this learning is not only the child's age characteristics, while students need to know is composed of a collection of different elements, which contains a collection of ideological content, so the in a simple counting process, slowly deepening the understanding of the basic knowledge of mathematics.

Understanding the numbers, the students began to learn the natural numbers, integers, fractions and decimals knowledge, in order to enable students to distinguish between the basic materials are generally used classification methods to help students to be distinguished. For example; which of the following number is an integer, which is the score? Look for, which of the following numbers is decimal, which is an integer? And so on. Through the integration of knowledge and job design, it is natural to a collection of modern mathematical ideas permeate them. When the students of this classification includes a collection of ideas, and then encounter a
similar situation, can use it to solve some conceptual confusion caused by the problem, greatly increased their understanding of basic concepts. Understanding of basic math and conventions common multiple number of teaching, teaching materials are usually given in the form of two charts from the public elected him their part, which is their common multiple or divisor. This collection reflects the thinking of the intersection of knowledge. In this form of teaching to allow students to find the greatest common divisor of two numbers, it is very clear, easy to understand and master students.

The Modern Mathematical Basis of Dialectical Thinking in the Integration of Mathematics Textbooks

In basic mathematics textbooks, solve word problems are different forms and types, but which are the dialectical integration of the mathematical idea that the contradiction between the known and unknown. For example, the distance problem is through the application of distance, velocity and time into the relationship between these three to be resolved, so the basic materials in there like "Distance = Velocity × Time" and other similar formulas, this expression that allows students to give the two known conditions, to look for another unknown number in order to solve the problem. In this process, students can build a model of their own problem-solving, learn how to solve the problem is the most appropriate and most easily understood. Therefore, this dialectical thinking in mathematics students stage show is necessary, it can exercise the students to identify problems, analyze issues and solve specific problems, while for students in logical thinking ability is also helpful.

Another dialectical thinking about mathematics performance on the basis of a simple mathematical equation in the content, such as One dollar equation is the equation of the relationship through to the unknowns of the equation to be solved, such as X +5 = 19, X-4 = 20 and other types of equations. Should be emphasized that equation method is no longer unknown to a known number and position of absolute, but also explicitly recognize the unknown number, recognize it and datum is the same. On the one hand, the equation is an equation containing the unknown, unknown is required; other hand, the equation also includes the solvability of this factor, which is known to side. It can be said to solve this equation is a process starting from the known conditions to the solution of the unknown process. So in the basic materials to begin the introduction of this simple equation, as is to make students understand and master are known to the unknown from the problem-solving process, rather than by simple arithmetic methods lengthy, step by step solution.

In the Mathematical Modeling of Materials in Modern Thinking

After the new curriculum, based on the reform of mathematics teaching has been greatly one hand, each volume has a mathematical angle, both to consolidate the knowledge and develop thinking. On the other hand, after the fourth grade has a copy of the final is closely related to modern life with the integrated application. These problems are related to life, first clear the
problem and gather information, modeling, discussion and analysis, or interpretation of the final solution. In the sixth grade "reasonable deposits topic" teaching, for example.

(1) **Clear Issues.** The activities are centered on: "Mom to deposit $10,000, for six years after college with his son how to gain the maximum deposit?" The problem started. The question contains several key messages: the principal can deposit period and the use of funds.

(2) **To Collect Information.** Specific problem, need to collect information related to this issue. Materials presented to the bank through counseling and access to relevant provisions of the way to get information: (a) RMB savings deposit interest rates, including regular Lump, installment savings, current interest rates. (b) Education savings deposit interest income tax exemption; it can save the time limit and the corresponding interest rate. (c) Bonds are also exempt interest income, a three-year and five-year......

(3) **Design.** According to the information collected to enable students to design a specific savings program. Regular savings deposit program can fill in the first 111 in the first table. Other deposit options, such as education savings bond program and the program can be bought to fill in the second table. Each requires a specific program to fill out a clear storage period, the interest due, interest due income taxes, and other information.

(4) **Options.** From the various possible options, select the largest gains, that is the optimal solution for a reasonable deposit, and calculate the total revenue.

**Conclusions**

Through the above analysis we can see the points, although some of the content of modern mathematics in basic education rarely directly material presented, but it contains some of the mathematical ideas that are embodied. These mathematical ideas for students to learn mathematics discipline is helpful, it has strengthened the students' understanding of the content of the basic knowledge and mastery on the other hand, the basis of these mathematical ideas and content integration, allow them to keep to solve mathematical problems into good habits, and gradually these ideas become a scientific problem-solving and structured of the rich, but also the mathematical essence of the discipline.

**References**


APPLICATIONS OF LINUX-BASED QT-CUDA PARALLEL ARCHITECTURE

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Abstract. Joint programming of QT and CUDA is a urgent problem on Linux, a Linux-based QT-CUDA parallel architecture has been built creatively. As an example, an fast parallel rendering algorithm for seismic and GPR imaging is proposed and implemented based on the Linux QT-CUDA parallel architecture. It is proved that the parallel rendering algorithm is about ten times faster than conventional algorithm, which can be widely applied to fast visualization of different kinds of 2D data.

Keywords: QT, CMAKE, CUDA, parallel computing, imaging.

Introduction

CUDA (Compute Unified Device Architecture)[1] parallel computing technology invented by NVIDIA in 2007 is the latest and powerful high-performance processing technology. The CUDA source code of GPU can only be compiled into an independent program on Windows so far. In order to achieve joint programming of QT [2,3] and CUDA, and take their advantages simultaneously, building a Linux-based QT-CUDA parallel architecture to achieve a high degree of QT and CUDA integration is of practical significance. After in-depth study and experiment, a QT-CUDA parallel architecture has been built creatively, which has solved the urgent problem of programming effectively. As an example, the fast GPU rendering algorithm for seismic and GPR (Ground Penetrating Radar) data visualization is proposed and implemented based on QT-CUDA parallel architecture. The GPU rendering algorithm is about ten times faster than conventional rendering algorithm on the same PC, which is proved to be the fastest seismic data visualization solution on PC currently.

A. QT Cross-Platform Features. QT introduced by Norwegian TrollTech is a cross-platform C++ GUI(Graphical User Interface) application development platform. It includes many features:

- Cross-platform feature supports Unix, Linux and Windows.
- Object-oriented feature and encapsulation mechanism make QT be a very high degree of modularity.
- Being rich of APIs(Application Programming Interface), including more than 250 classes.
- Using private signals and slots technology to communicate between objects.
- supporting 2D and 3D graphics rendering.
QT implements communication between objects by means of signals and slots technology. For example, an occurrence of event can fire one or more signals; any object can define slot functions to make different responses. Signals and slots technology has completely overcome two major shortcomings of C++ callback method. Besides, all slots are executed by means of multi-threads and asynchronous. The relation between signals and slots is loose. Those features make QT being a good choice of client-server software and embedded software development tools. But QT itself does not have the parallel processing capability, it must be integrated with CUDA to carry out.

B. GPU Parallel Principle [4]. In 1999, NIVDIA released the first GPU graphic processors, GPU hardware maintains a rapid development by now, GPU performance can double every year, much higher than CPU (Central Processing Unit), which follows Moore's Law of every 18 to 24 months to double the performance. However, the GPU parallel computing can not be carried out until the official release of CUDA by NIVDIA in 2008. GPU-based parallel techniques can be simply described as following: CPU does data preparation, and executes serial initialization code; GPU carries out parallel computing. GPU-based graphic processor directly connects to CPU by AGP or PCI-E bus, which owns independent high-speed memory and lightweight threads. GPU can achieve zero-overhead fast thread switching. At present, NIVDIA's GPU contains more than 1 to 30 SMs (Stream Processors), which is equivalent to 1 to 30 CPUs. Each SM has eight SIMDs (Single Instruction Multiple Data), which is equivalent to 8 CPUs cores. And a SIMD has a thread block of up to 512 threads. Therefore, the performance of GPU can be 10 times faster than performance of CPU at the same period. Besides, a host (computer) can install multiple GPU graphic cards; therefore, the performance of GPU parallel computing on PC can reach high performance of the cluster computing. CUDA parallel programming model is shown in Figure 1[1], which comes from the NVIDIA CUDA programming guide. In Figure 1, CPU is on behalf of Host executes Serial Code (serial code program); Device is on behalf of GPU executes parallel kernel functions (parallel kernel code); Grid is on behalf of abstract stream processors grid and Block is on behalf of an abstract grid of threads.

Implementation of QT-CUDA Parallel Architecture

At present, CUDA can only use C programming language, which supports a little C++ syntax, so it can not be combined with QT programming. It is very urgent to solve the problem of integrating QT and the CUDA programming. By in-depth study of the existing open source software CMAKE and joint programming technique, an innovative QT-CUDA parallel architecture is presented and implemented on Linux box after repeated tests. The steps as follows:
A. Setup the Latest Open Source Packages (QT, CUDA, and CMAKE) on Linux System properly. Among them, CUDA carries out parallel processing of GPU programming; QT carries out serial code programming and calling CUDA parallel processing function; CMAKE coordinates the link between QT and CUDA, generating Makefile for Gcc compiler to compile codes automatically. The three divisions together constitute the QT-CUDA parallel architecture.

B. Create Subdirectories for QT-CUDA Parallel Architecture. According to directory structure shown in Figure 2, create a QT-CUDA parallel development subdirectory named qt-cuda, and design CMAKE main configuration file CMakeLists.txt. The codes of CMakeLists.txt[5] are as follows:

```bash
# specify the target project name
PROJECT (target)

# Cmake specify CMAKE version 2.6.0 or higher
CMAKE_MINIMUM_REQUIRED (VERSION 2.6.0)

# QT4 specify the QT 4.0 or higher version
FIND_PACKAGE (Qt4 REQUIRED)

# CUDA specify the CUDA parallel software
FIND_PACKAGE (CUDA REQUIRED)

# Source specify the source subdirectory of all codes
subdir SUBDIRS (codes)
```

Fig. 1 CUDA parallel programming model.
C. Design Another Configuration File Cmakelists.Txt And Place It At Directory Of Codes, Put All QT Source Code At Subdirectory Of Qt, And Put All CUDA Source Code At Subdirectory Of Cuda. The codes of CMakeLists.txt are as follows:

```plaintext
# setup qt source code
SET (QT_USE_QTOPENGL 1)
SET (QT_USE_QTXML 1)
INCLUDE (${QT_USE_FILE})

# list all qt program files here
SET (qt_cuda_SRCS_CXX qt/*.*.cpp)

# list all qt head files here
SET (qt_cuda_MOC_HDRS qt/*.*.h)

# list all qt GUI ui files here
SET (qt_cuda_UIS qt/*.*.ui)

# list all cuda parallel program files here
SET (qt_cuda_CUDA_SRCS cuda/*.*.cu)

# specify method for compiling qt ui files.
QT4_WRAP_UI (qt_cuda_SRCS_CXX $ {qt_cuda_UIS})

INCLUDE_DIRECTORIES
($ {CMAKE_CURRENT_BINARY_DIR})

# specify method for compiling qt moc files
QT4_WRAP_CPP (qt_cuda_SRCS_CXX
$ {qt_cuda_MOC_HDRS})

# specify the joint compiling method for qt and cuda codes
CUDA_ADD_EXECUTABLE
({${CMAKE_PROJECT_NAME}}
{Qt_cuda_CUDA_SRCS}$
{qt_cuda_SRCS_CXX})TARGET_LINK_LIBRARIES
```

Fig. 2 Directory structure of QT-CUDA parallel architecture
($ \{\text{CMAKE\_PROJECT\_NAME}\} \{\text{QT\_LIBRARIES}\})
INSTALL (TARGETS $\{\text{CMAKE\_PROJECT\_NAME}\}$ DESTINATION bin)

D. Write CUDA Source Codes. By coding with CUDA, main function can not be defined. Only
the normal functions and the parallel processing kernel functions can be defined.

E. Define and Execute CUDA Functions in QT Files. Declare external CUDA function in QT
using ‘extern “C” void cuda_function_name (parameter1, parameter2)’, and execute CUDA
parallel processing function by passing real values to all parameters. The cuda_function_name is
a CUDA function’s name; parameter1, parameter2 are parameters of cuda_function_name.

F. Create a Linux Bash Shell Script File Named Linux_Build.sh for Finishing Compilation
automatically. The file named linux_build.sh is placed at subdirectory of qt-cuda. When
linux_build.sh is executed, all QT and CUDA files can be automatically compiled into a single
executable file with no errors. After the QT-CUDA parallel architecture is established, it is very
simple to add or remove QT and CUDA files by manually modifying the configuration file
CMakeLists.txt in subdirectory of codes.

Application of QT-CUDA Parallel Architecture
The color image of seismic data in oil and gas exploration is a very important tool to demonstrate
the result of seismic data processing and interpretation. A regular rendering algorithm can cause
GUI frozen and other problems [2,6] due to slow response on PC. In this thesis, an innovating
and fast parallel rendering algorithm is proposed and implemented based on above QT-CUDA
parallel architecture, which applies the latest GPU parallel technology to optimize rendering
performance by GPU multi-threaded parallel processing and then directly writing OpenGL
mapping buffer. It is proved that the parallel rendering algorithm is about ten times faster than
conventional algorithm. The GPU parallel rendering algorithm flow chart is shown in Figure 3.
The GPU parallel processing kernel function code is given as follows:

```c
__global__ void prepareMap (float3 * positions, float * gp_r, float * tmp_color, int screenWidth, int screenHigh, float min, float max, int colorTableLength) {
    unsigned int i = blockIdx.x * blockDim.x + threadIdx.x;
    unsigned int j = blockIdx.y * blockDim.y + threadIdx.y;
    if (i < screenWidth && j < screenHigh) {
        int index = (int)((colorTableLength - 1) * (gp_r + i * screenHigh + j) / (Max - min) - (colorTableLength - 1) * min / (max - min));
        if (index < 0) index = 0;
        if (index > colorTableLength - 1) index = colorTableLength - 1;
        positions[j * screenWidth + i] = make_float3(*(tmp_color + index * 3 + 0), *(tmp_color + index * 3 + 1), *(tmp_color + index * 3 + 2)); //generate pixel data of map
    }
__syncthreads (); //multi-thread synchronization
}
```

Due to space limitation, the main codes of OpenGL initialization, generating pixel data and writing OpenGL buffer are only given as follows:

```c
glMatrixMode(GL_PROJECTION); //OpenGL initialization
 glLoadIdentity();
 glRasterPos2f(-1.0, -0.85); glScalef(1.0, 1.0, 1.0);
 glViewport(0, 0, screenWidth, screenHigh);
 for(int j = 0; j < screenHigh / 2; j++) //regulating pixel data
 for(int i = 0; i < screenWidth; i++) {
```

---

**Fig. 3 Flow chart of seismic imaging parallel rendering algorithm**
float3firstData=*(positions+j*screenWidth+i);
*(positions+j*screenWidth+i)=*(positions+(screenHigh-2-j)*screenWidth+i);
*(positions+(screenHigh-2-j)*screenWidth+i)=firstData;
// writing OpenGL buffer

glDrawPixels(screenWidth,screenHigh,GL_RGB,GL_FLOAT,positions);

The seismic imaging [7] by GPU parallel is illustrated in Figure 4. With the means of the parallel seismic data rendering algorithm, fast imaging can be carried out at any processing and interpretation stage. “What you see is what you get” [8] can be carried out with no GUI frozen. The oil reservoir at blue polygon area is clearly shown at the centre of Figure 4. The same rendering algorithm is applied to implement GPR imaging and achieves good results [9,10]. Figure 5 is a fast imaging of underground pipe GPR field data based upon GPU imaging parallel rendering algorithm. Imagings are finished on the laptop Acer ASPIRE 4736G (containing GE Force G105M CUDA 512M graphics card). With regard to 1280×768 resolution image, the GPU imaging parallel rendering algorithm can get speedup of 10 respectively.

![oil reservoir](image)

Fig. 4 Seismic fast imaging of an oil field
Conclusions
A GPU parallel computing technology is the latest and powerful high-performance processing technology. However, the current CUDA source code can only be compiled into an independent executable program. A QT-CUDA parallel architecture has been built creatively on Linux, which has solved the urgent problem of programming effectively. At last, the seismic and GPR imaging is carried out by a fast GPU rendering algorithm based on QT-CUDA parallel architecture. This solution can also apply to fast visualization of 2D data.

Acknowledgement
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References


CONTOURLET TEXTURE RETRIEVAL FILTER CHARACTERS

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Abstract. Contourlet transform has been proved superior to wavelet in representation piecewise texture and contour characters in digital images. In order to improve the retrieval rate further, we studied the representation abilities of some different filters for texture features in two contourlet retrieval systems. The two systems both used Canberra distance for similarity measure, but different feature vectors which composed of energy combined kurtosis and energy combined standard deviation, respectively. Experiment on a 109 texture image database from MIT results show that the filter “pkva” performs better than other ones including bi-orthogonal 9-7, haar, burt, bi-orthogonal 5-3, etc, for the two retrieval systems, under the same system structure.

Keywords: content based image retrieval, contourlet transform, texture image, retrieval system, and filter.

Introduction

How can we tell the difference between different types of objects? Maybe several characters can be listed including texture, shape, color, etc. Obviously, texture is a very important character. We always found them in all kinds of application fields including fingerprints, iris, feathers, clothes, leaves, material, zebra, worm, etc. Man must distinguish these different kinds of patterns in our daily life and research work.

In many situations, we must find something similar of the surface characters; maybe, texture is the important feature at some time. The procedure is often called texture retrieval. The classic method maybe search them in a database by keywords or some natural language describe, this work style is suitable for noted carefully image database. To overcome the difficulties of keyword retrieval systems, a new type of retrieval system called content-based image retrieval (CBIR) system was proposed[1-2]. The most important technology of the CBIR system includes two aspects: feature extraction and feature comparison. During the past decade years, wavelet transform has played an important role in the system due to its good characters of multi-scale and local time-frequency [3-4].

Yet, some disadvantages of wavelet transform including shift sensitivity and the lack of directionality limits its abilities in texture representing. To overcome the deficiencies of wavelet transform, researchers have developed many improved approaches, including contourlet transform (CT) [5]. Since the transform was proposed in 2002 by Do, several modified versions
have been proposed and form a new family including non-subsampled contourlet transform (NSCT) [6] and localized contourlet version [7], etc. Non-subsampled contourlet transform which was proposed by Cunha in 2005 has higher shift insensitivity level than the original contourlet transform but has higher redundancy, the high redundancy makes the transform much more time consuming and much larger memory needed.

To overcome the limitation of high redundancy, Cunha presented a compromise transform which was a cascade of non-subsampled Laplacian pyramid and critical subsampled directional filter banks, and made the redundancy fall to S+1, where S is the decomposition scale in the transform. Here we call the transform Contourlet-S.

To further reduce the redundancy of the transform and make sure that the transform anti-aliasing and shift insensitive, Yue Lu modified the Laplacian pyramid filters in the band transition and subsampled rate, and followed by the original directional filter banks, got more localized time-frequency characteristics [8]. According to the difference of the redundancy, the new transform can be classified into 3 different versions named Contourlet-1.3, Contourlet-1.6 and Contourlet-2.3. Because the new version Contourlet-1.3 utilized the full critical subsampled filter banks, its redundancy equals that of the original vision which was proposed by Minh. Do. Ever since the contourlet transform was proposed, many literatures reported the application approaches in many different areas including CBIR systems. We have constructed some texture image retrieval systems based on the above different versions of contourlet transform [8-10].

From statistical view, Wouwer et al. employed generalized Gaussian density functions to represent texture images in the wavelet domain [11]. The model parameters are estimated using a method of moment matching (MM), and the similarity function is defined as weighted Euclidean distances on extracted model parameters. Do and Vetterli developed a new frame that combine FE and SM together and got a promising result in wavelet-based texture image retrieval rate comparing with traditional methods [2]. Yang extended this idea to contourlet domain [12].

There are several other statistical values for the sub-bands coefficients, including mean, energy, skewness, and kurtosis, etc. And, each of them can be used as retrieval feature. In our recent work, we have incorporated kurtosis, with energy, that is, using their combination to build feature vector, survey their abilities in contourlet retrieval system and show that the new combination feature make significant improvement in retrieval rate than the traditional approaches under Canberra distance measure metric.

The remaining parts of this paper are organized as follows: feature vectors and similarity measure will be covered in section 2, experimental method and results will be shown in section 3, in the section 4, the last section, we will conclude the whole paper.
**Contourlet Retrieval System Used for Discussion**

We should introduce the system briefly here for Integrity, for detail, we recommend the paper we have published [13].

A. **Feature Vectors.** For each kind of wavelet or contourlet-like texture image retrieval system, we should first transform the space domain image into wavelet or contourlet domain. The image data in destination domain is composed of several sub-sets always called sub-band or sub-channel due to its some local characters and transfer situations.

For a sub-band in contourlet domain, we use Eq. 1 to calculate its energy, where \( W(\bullet) \) stands for coefficients in contourlet domain, \( E(s,k) \) denotes the average energy of the band which is indexed by scale \( s \) and direction \( k \), and \( M,N \) stand for the row and column number of the sub-band coefficients.

\[
E(s,k) = \frac{1}{MN} \sum_{m=1}^{M} \sum_{n=1}^{N} | W_{s,k}(m,n) | 
\]

(1)

The kurtosis is shown in Eq. 2 and standard deviation defined as Eq.3, where \( M,N,s,k \) have the same meaning as in Eq. 1, \( \sigma(s,k) \) means the standard deviation of a certain sub-band coefficients, \( \mu_{s,k} \) denotes the average value of the sub-band coefficients.

\[
\Gamma(s,k) = \frac{1}{MN} \sum_{m=1}^{M} \sum_{n=1}^{N} \left( W_{s,k}(m,n) - \mu(s,k) \right)^2 
\]

(2)

\[
\sigma(s,k) = \left[ \frac{1}{MN} \sum_{m=1}^{M} \sum_{n=1}^{N} \left( W_{s,k}(m,n) - \mu_{s,k} \right)^2 \right]^{1/2}
\]

(3)

In this paper, we survey the filter characters for retrieval application in two different retrieval systems. In the first system, each feature vector is constructed by cascading energy and kurtosis of each contourlet domain directional sub-band, for convenience, we call it system1. In another, each feature vector is combined by cascading energy and standard deviation of each contourlet domain directional sub-band, for convenience, we call this one system2. For every image in the database which will be retrieved, a certain feature vector can be obtained and then is put into the feature vector database as the signature of the corresponding image for retrieval.

B. **Determination of Similarity Measure.** The similarity measure is used to calculate the distance between different feature vectors. Up to now, at least there are 10 different types of distance measure, they are: Manhattan (L1), Weighted-Mean–Variance (WMV), Euclidean (L2), Chebychev (L), Mahalanobis, Canberra, Bray-Curtis, Squared Chord, Squared Chi-Squared and Kull-back Leibler. Kokare compared the nine measures except Kull-back distance (KLD) and declared that Canberra and Bray-Curtis are superior to others, and we compared Canberra and Kull-back distance, the result is that Canberra is more suitable in such kind of situation. So in this paper, we directly choose Canberra distance as distance measure. The Canberra distance is
defined as formula (4), where \( d(x, y) \) means the distance between vector \( x, y \), \( D \) denotes the dimension of the feature vectors, \( x_i, y_i \) are the i-th components of \( x \) and \( y \), respectively.

\[
d(x, y) = \sum_{i=1}^{D} \frac{|x_i - y_i|}{|x_i| + |y_i|}
\]  

(4)

C. Filters for Comparison. There are many different types of filters which can be used for constructing the frame of contourlet transform and its different variations. For clarity, we select some common used ones for comparison. The filters will be compared in this paper include “pkva”, “McClellan transformed of 5-3 filters”, “McClellan transformed of 9-7 filters”, “bi-orthogonal 9-7”, “bi-orthogonal 5-3”, “haar”, “burt”, etc.

Experiments and Results

We will introduce the experimental objects and results in this section.

A. Experimental Objects. The experimental objects are the 109 texture images come from Brodatz album [14]. For each 640×640 pixels image, we cut them into non-overlapped 16 sub-images and each one is 160×160 pixels size, then we can obtain an image database with 109×16=1744 sub-images. The 16 sub-images come from the same original image can be viewed as the same category.

B. Experimental Results. The average retrieval rate can be described by Eq. 5, where \( q=1744 \), \( R(p) \) denotes the average retrieval rate for each \( p \in \{16, 20, 30, 40, 50, 60, 70, 80, 90, 100\} \), hence 10 retrieval results can be acquired. \( S(p, i) \) is the number of images belong to the correct group when the i-th image used as query image.

\[
R(p) = \frac{1}{q} \sum_{i=1}^{q} R(p, i) = \frac{1}{q} \sum_{i=1}^{q} \frac{S(p, i)}{16}
\]  

(5)

Using the above approach, we can get the average retrieval rate of the contourlet texture image retrieval system using different filters as shown in table 1 and table 2, for the two different retrieval systems. In the two tables, the first column vector identifies the filter type of Laplacian pyramid and directional filter respectively.

As for the decomposition umbers of scale and directional filter bank, we used [4 3 3] for this experiment, which means that we decompose the images at three scales and from fine to coarse scale, 16,8,8 directional subband for each scale, adding the low frequency sub-band, the number of sub-bands are 33, each sub-band needs two parameters to describe, so, for every sub-image in the database, the dimension of feature vector is 66. Using the same method for every sub-image, we can extract 1744 feature vectors altogether. All the feature vectors were put together into feature vector database. For the experimental data in this paper, the size of feature vector database is 66×1744=115,104 Bytes (excluding the structure information). It should be noted that the size of the feature vector database is much smaller than the original sub-image database which size is 44,646,400 Bytes.
From table 1, we can see that “pkva” has the most significant ability in texture representation. Using “pkva” as Laplacian pyramid and directional filter can perform better than the other filters, especially under small p values which is a good character for such system because in practical situations, we always don’t want to improve retrieval rate by increasing the value of p. From table 2, we also find that “pkva” is superior to other filter types; the conclusion is very similar to the case in table 1.

We examined many other decomposition parameters on the two retrieval systems, and the similar results can be obtained. It should be noted that NO-Free-Lunch law do exist here. Although “pkva” has the advantages of highest retrieval rates, the most notable disadvantage is that time consuming much more that other kinds of filters due to its big size of coefficients.

Table 1 Comparison of retrieval rates using different filters for system1 (%)

<table>
<thead>
<tr>
<th></th>
<th>1</th>
<th>2</th>
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<th>4</th>
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<td>89.3</td>
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<td>80.5</td>
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<td>79.8</td>
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<td>88.8</td>
<td>89.6</td>
<td>90.3</td>
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<td>91.1</td>
</tr>
</tbody>
</table>
Conclusions

Some filter characters in texture representation has been compared for two different contourlet retrieval systems. Experimental results tell us that the filter “pkva” is excellent. It should be noted that we have done some similar experiments on other contourlet like retrieval systems including non-subsampled contourlet transform, contourlet-2.3, contourlet-1.3, dual-tree complex contourlet transform, etc., and almost the same results can be gotten. The next question we should consider is how to shorten the “pkva” filter, while containing its excellent character or construct a new filter superior to it.

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References


FAULT DIAGNOSIS FOR SHIP’S ANTI-ROLLING SYSTEM BASED ON BP-FNN

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Abstract. According to a certain model ship’s anti-rolling system, this paper analyzes the fault information and establishes a fault diagnosis model using fuzzy-nerve network algorithms. Based on the fuzzy logic processing data and using the past experience and knowledge, the nerve network avoids some problems of fault tree diagnosis system, such as matching conflict, combination explosion, and infinite recursion. In order to train the nerve network, this paper adopt the improved BP arithmetic which can solve the problem of convergence speed and convergence surge. The result shows this fault diagnosis system has strong robustness and generalization. That method that uses model free diagnosis is easy to learn by itself and constant perfect system function, and has some theories and engineering application value.

Keywords: fault diagnosis, neural network, fuzzy logic, anti-rolling system.

Introduction

Ship’s anti-rolling system can reduce ship-rolling effectively which composed by complex machine, electricity and liquid system. It cost a lot of manpower and time to check trouble when system works abnormally. So strengthening the fault diagnosis on ship’s anti-rolling system is a key to ensure equipment operation safety. The diagnosis technique can make early prediction of system fault developing process, find the trouble reason in time and present corresponding measures to avoid the occurrence of the trouble.

Fuzzy logic, which simulates the brain’s logical thought, has obvious logicality and transparent. The model has clear physical conceptions, its structure parameters has clear physical concept, and it can deal with explicit logical issues, trace and resolve the system’s inference procedure and conclusions easily[1]. For the complex structure, however, the time of law establishment and inference show exponential growth with rule numbers. It is hard to establish fuzzy relation matrix, the accuracy is poor and the application of fuzzy logic system is restricted.

Fuzzy Neural Network is a new network which absorbs the advantages of fuzzy theory and neural network and makes up their own drawbacks. Duan[4] used the FNN’s property of function approximation, so as to resolve the problem of learning state space is too large. Wang[5] introduced a non-linear momentum term in BP algorithm, and combined with heating strategy in order to train FNN rapidly and globally. On those basis, this paper has designed a FNN model, Taking a certain type of ship’s anti-rolling system as an example, use fuzzy theory to processes fault feature parameters, and combine with the improved BP neural network to
diagnose fault, the result shows the method is effective, and has certain value of theoretical guidance and engineering application.

**Fault Analysis of Ship’s Anti-rolling System**

Ship’s anti-rolling system is a complex and mechanical-electrical-hydraulic closed loop control system. The relation between fault symptom and reason is complicated. At present, the fault diagnosis of anti-rolling is mainly based on the experts’ knowledge and operators’ experiences.

From the above analysis, there are so many problems on using traditional method of fault tree to diagnose fault. Firstly, for the relationship between the omen and the fault is complex, it is quite difficult to establish suitable fault tree. Secondly, given established fault tree, it would be hard to update fault tree, the relationship between the omen and the fault would change with time of system’s tenure use. Thirdly, it is inconvenience for users. And the fatal is that could not match. This paper tries to apply artificial intelligence algorithm to approach the nonlinear mapping relationship between the omen and the fault in order to resolve the problem of ship’s anti-rolling system’s fault diagnosis.

**Fuzzy Neural Network Structure**

Fuzzy Neural Network is composed of input fault symptom fuzzy processing and neural network learning training. It contains four layers. The first one is fuzzy fault omen input layer, which transforms the input of sensor into values of membership degree. The second one is neural network training samples input layer, which is a fully connected structure. The third one is hidden-layer neural network. And the forth layer is the output of diagnosis fuzzy neural network system, which produces the diagnosis results in the form of probability, and guides maintainers do testing work.

**Fuzzy Neural Network Algorithm Fuzzy**

**A. The Fuzzy Neurons.** The fuzzy neurons are used to get membership degree’s values of input fault omen eigenvalue, and then translate sensors-detect value into fuzzy volume. It is assumed that \( h \) premonitions would be found in diagnosis object, which is expressed as follows: \( x_1, x_2, \ldots, x_h \). Function can be transferred to fuzzy valued \( \mu_1, \mu_2, \ldots, \mu_h \) through fuzzy layer, where fuzzy function is generally Gaussian Function. Gaussian has good smoothness and precision concept, so as to ensure membership function and proper symmetrical balance and agrees with language order. The overlap is seldom between functions.

\[
\mu_A(x) = e^{-\frac{(x-a)^2}{2\sigma^2}}
\]  

(1)

**B. BP Neural Network Algorithms.** Multilayer feedforward neural network adopted BP(Error Back Propagation) algorithm. Further, Robert Hecht Hielson pointed out if only there were enough nodes, that can use arbitrary precision approaching a nonlinear function in the BP
topological structure of neural network, even though there was only one hidden layer in neural network. In fact, the model of fault diagnosis uses BP’s function approximation ability to approach fault clustering boundary, so as to implement the nonlinear mapping from feature space to fault space.

C. Improved BP Neural Network Algorithm. In order to improve the learning speed of BP neural network and avoid the oscillating, this paper has also adjusted learning coefficient $\eta$ dynamically. According to the changed value of iteration total error through connection weight value adjusted between time $t$ and $t-1$, calculate the equation $\Delta E(t, t-1) = E(t) - E(t-1)$, and then judge the effectiveness of the adjustment about time $t$ by the plus-minus of $\Delta E(t, t-1)$. If $\Delta E(t, t-1) > 0$, it is effective, otherwise, ineffective. According to the formula (4) adjust the corresponding learning coefficient $\eta$.

$$\eta(t) = \begin{cases} \eta(t) + \beta \eta(t) & \Delta E > 0 \\ \eta(t) - \beta \eta(t) & \Delta E \leq 0 \end{cases}$$

(4)

Where, $\beta \in (0, 1)$, is the learning adjustment coefficient.

5. Software Design of Fuzzy Neural Network. The FNN algorithm is compiled by using the software Matlab7.0. It can transform the input variable of fault omen into the fuzzy membership of neural input variables through connection weight coefficient of fixed fuzzy layer. Network-learning adopts the improved BP algorithm. Figure 1 shows the flow chart of FNN algorithm.

![Fig. 1 Flow chart of FNN algorithm](image-url)
Example of Fuzzy Neural Network Diagnosis

In order to check the effectiveness of default diagnosis system based on fuzzy neural networks designed above, this research tested it as a certain type ship’s anti-rolling system. According to the principle of anti-rolling system, the experts’ knowledge and the specific fault cases, we choose some typical faults as follows, \( f_1 \)-the sensor breakage in angular velocity of roll, \( f_2 \)-the fault of PID controller, \( f_3 \)-the fault of ship speed regulator, \( f_4 \)-the fault of Servo System; \( f_5 \)-the fault of angle-feedback potentiometer.

Train of Fuzzy Neural Network. In order to extract the experts and maintainers’ experience of fault diagnosis, it is necessary to fuzzy up the fault case. Each fault symptom signal has been fuzzed as three grades, “small, S”, “normal, N”, and “large, L”. According to the possibility, the fault reason has been fuzzed as five grades, I, II, III, IV, and V where the higher the grade is, the more possibility the fault is of. On the basis of experts and maintenance’s experience, we have established the fuzzy rule sets, shown as Table 1.

<table>
<thead>
<tr>
<th>Portent signal</th>
<th>Fault Reason</th>
</tr>
</thead>
<tbody>
<tr>
<td>( U_{o} )</td>
<td>( D_{o} )</td>
</tr>
<tr>
<td>S</td>
<td>S</td>
</tr>
<tr>
<td>N</td>
<td>S</td>
</tr>
<tr>
<td>N</td>
<td>N</td>
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<tr>
<td>S</td>
<td>N</td>
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<td>N</td>
<td>N</td>
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<td>N</td>
<td>N</td>
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<tr>
<td>S</td>
<td>L</td>
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<tr>
<td>N</td>
<td>L</td>
</tr>
<tr>
<td>L</td>
<td>L</td>
</tr>
</tbody>
</table>

When the ship speed is 18kn, the normal output voltage amplitude of the ship speed sensitivity regulator, \( U_{o} \) is 9V, while \( U_{o} = 11V \) is too large, either \( U_{o} = 7V \) is too small. Using root mean square of fin as fault characteristic parameters, denoted by \( D_{o} \), \( D_{o} = 5 \) is normal, while \( D_{o} = 7 \) is too large, either \( D_{o} = 1 \) is too small.

According to the experts’ experience, the fuzzy quantities I, II, III, IV and V in the table 1 are valued 0.1, 0.3, 0.5, 0.7, 0.9 respectively, and then transforme the fuzzy rule sets into training sample set of neural network, shown as Table 2.
Table 2 The training sample of fuzzy neural network

<table>
<thead>
<tr>
<th>Portent signal</th>
<th>Fault Reason</th>
</tr>
</thead>
<tbody>
<tr>
<td>$U_o/V$</td>
<td>$D_o/V$</td>
</tr>
<tr>
<td>7</td>
<td>1</td>
</tr>
<tr>
<td>7</td>
<td>1</td>
</tr>
<tr>
<td>7</td>
<td>1</td>
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<tr>
<td>7</td>
<td>1</td>
</tr>
<tr>
<td>7</td>
<td>1</td>
</tr>
</tbody>
</table>

The connection weight of the fuzzy layer has decided the quantization level of input data.

Using the sample data in the table 2, we train and learn the fuzzy neural network repeatedly until the error precision result had been accepted. Where error precision, learning coefficient, momentum factor, and coefficient of learning adjustment were set respectively as, $E_{set} = 1 \times 10^{-10}$, $\eta = 0.65$, $\alpha = 0.1$, $\beta = 0.02$.

**Fig. 2 Surface of FNN training specimen error.**

**B. Example of Fault Diagnosis.** After implement training, the fuzzy neural network can be applied to actual diagnostic test. Input $U_o$ and $D_o$ which collected by sensor to network, output
the probability value of diagnosis result $f_1, f_2, f_3, f_4$ and $f_5$. Also, according to the diagnosis result, maintainers can debug one by one, which save a lot of fault detection time on the complex system, and lower requirements on the maintainers’ skill. Input the given malfunction omen eigenvalues of anti-rolling system into fuzzy neural network then obtain the fault diagnosis result, shown as Table 3.

<table>
<thead>
<tr>
<th>Portent signal</th>
<th>Fault Reason</th>
</tr>
</thead>
<tbody>
<tr>
<td>$U_o I$</td>
<td>$D_o I$</td>
</tr>
<tr>
<td>7.8</td>
<td>5.1</td>
</tr>
<tr>
<td>8.7</td>
<td>2.2</td>
</tr>
<tr>
<td>9.5</td>
<td>8.4</td>
</tr>
<tr>
<td>9.0</td>
<td>5.0</td>
</tr>
</tbody>
</table>

It is observed from Table 3 the fault diagnosis system has strong abilities of robustness and generalization. All the diagnosis result is correct and provide guidance for the maintainers. Furthermore, maintainers can accumulate fault diagnosis cases on training fuzzy neural network, improve system performance continuously.

**Conclusions**

The ship’s anti-rolling fault diagnosis system based on fuzzy nerve network this paper designed has very good diagnosis result. On the one hand, it diagnoses without models instead of precise mathematical model. The explicit structure is excellent and the physical meaning is explicit. On the other hand, the system has self-learning function, strong robustness and generalization. The fuzzy neural network fault diagnosis system is an open platform, which takes full advantage of previous experience, accumulates maintenance experience continuously, and improves the fault diagnosis ability. This study has certain theories and engineering application value.

**Acknowledgement**

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**References**


AUTOMATIC TESTING SYSTEM FOR EMC TESTING WITH WAVELET ANALYSIS APPLICATION BASED ON LABVIEW

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Abstract. According to the problem that the speed of electromagnetic compatibility (EMC) manual system is slow, a automatic testing system is designed for power supply line conducted emission based on general-purpose interface bus (GPIB) and LabVIEW, which actualizes a series of functions, such as signal generating, monitoring, analyzing and showing. It introduces the constituting of system, testing principle, software structure, and solves the hard problems and key techniques such as hardware’s controlling, algorithm’s fast convergence, compatibility of software. Because the testing signal include the ambient noise, the wavelet analysis is applied to filter the noise and gain the real conducted electromagnetic interference. Comparing with the other traditional apparatuses and the manual testing, the system improves the test efficiency and has good extensibility.

Keywords: Electromagnetic Interference, LabVIEW, Automatic Testing System, Wavelet Analysis.

Introduction
With the development in new technology of weapon, the number and type of highly sophisticated electronics equipments and system fitted in the congested shipboard, aircraft, guided missile environment, so electromagnetic environment (EME) is extremely complicated. Some equips were effected by electromagnetism energy in our practice. Therefore, this is an exigent and necessary mission of solving electromagnetic compatibility (EMC) problem. The technology of EMC is not only a new and integrative subject domain, but also a application of engineering scientific technology. As T. Shinuzuka depicted in [1], the study of EMC technology and solving EMC problem base on a lot of test research and measure in practice. We need to evaluate EMC of equip and find electromagnetic interference (EMI) problems by test. The problem is presented detailed by B. Sreedevi and N. S. Harischandra Rao in [2].

With the electrical equipment and systems on the modern ship increasing quickly, communication between the equipment is frequent. At present, most EMC testing is manual and the efficiency is poor, so it becomes an urgent task that testing the EMC system fast and reliably. Gong Fengxun and Ma Yanqiu introduced a universal estimating and measurement method of EMI[3]. By researching the EMC testing methods and EMC technique, the paper designs a automatic testing system which can test, save, analyse and process data. Because the testing signal include the ambient noise, the wavelet analysis is applied to filter the noise and gain the real conducted electromagnetic interference. Because of virtual instrument introduction, the
system has short development cycle and high operation efficiency. There is an automated test system for EMC already in existence which is called TILE (Totally Integrated Laboratory Environment System). The paper designed the system which can also undergo "digital signal processing" with wavelet analysis application. At the same time, it has strong extensibility and repeatability which can reduce repeated investment. The research results have important meanings on advancement of testing methods, quality of EMC testing and improvement of EMC techniques. The system has great value to popularize and application, also achieve innovation achievements on hardware drivers based on LabVIEW. Through reading and writing equipment drivers, it can make the equipment collect, analyse and save the signal. At last, the system will give a detailed report and a frequency graph.

Wavelet Analysis

There are much localization in the nonstationary signal analysis with Fourier Transform due to its resolution which Charles K Chui studied. The problem of the detection of a transient signal of unknown amplitude and arrival time, which is buried in noise, is not restricted to electromagnetic signal, but is general to many practical situations. The wavelet analysis is a powerful tool for nonstationary signal analysis because of the better resolution in frequency and time domain. Since the nonstationary signal has complex frequency component, the wavelet transform can set different resolution for noise filtering. Hao Zhang, T.R. Blackburn, B.T. Phung and D. Sen used wavelet transform technique for On-line Partial Discharge Measurements[4]. Yue Zhao, Niu Wencheng presented the application of wavelet analysis in ultrasonic sensor system characteristic signal pretreatment[5].

A.Wavelet Transform. For the function \( \phi(x) \in L^2(R) \), the subspace \( V_j \) is generated by \( \phi_{j,k}, k \in Z \), just as follow:

\[
V_j = \text{span}(\phi_{j,k}, k \in Z), \quad j \in Z
\]  

(1)

In order to construct the model of wavelet analysis, some definitions is given:

Definition 1: the space series \( \{V_j\}_{j \in Z} \) in the space \( L^2(R) \) meets the follow conditions:

(1) uniform monotone: \( \cdots V_{-1} \subset V_0 \subset V_1 \cdots \)

(2) gradual complety: \( \bigcap_{j \in Z} V_j = \{0\} \)

(3) expansion in rule:

\[
f(x) \in V_j \Leftrightarrow f(2x) \in V_{j+1}, \quad j \in Z
\]

(4) Riesz base exists: there is \( \phi \in V_0 \), which makes that \( \{\phi(x-k)\}_{k \in Z} \) is the Riesz base between limit A and limit B, the base is as follow:

\[
\phi_{j,k}(x) = 2^{j/2} \phi(2^j x - k)
\]

(2)
where $\forall j \in \mathbb{Z}, \{\phi_{j,k}, k \in \mathbb{Z}\}$, $\phi$ is scale function.

Then we call Multiple Resolution Analysis (MRA) in the space $L^2(R)$.

If $\phi$ generate one MRA, because $\phi \in V_0 \subset V_1$, and $\{\phi_{j,k}, k \in \mathbb{Z}\}$ is a Riesz base in $V_1$, then we get the relationship between scale functions $\phi$ in $\{p_k\}$

$$\phi(x) = \sum_{k=-\infty}^{\infty} p_k \phi(2x - k) \quad (3)$$

Given that the wavelet meets the permission, the wavelet in $V_1$ can be generated as follow:

$$\psi(x) = \sum_{k=-\infty}^{\infty} q_k \phi(2x - k) \quad (4)$$

Function family $\{\psi_{0,k}\}$ generates a close subspace $W_0$ as follow

$\psi_0 = \text{span}\{\psi_{0,k}, k \in \mathbb{Z}\} \quad (5)$

Therefore, $W_0 \subset V_1$ from (2) and (5). When construction wavelet, guarantee that $V_1$ is the direct sum of $W_0$ and $V_0$, that is

$V_1 = V_0 + W_0 \quad (6)$

Definition 2:

$W_j = \text{span}\{\psi_{j,k}, k \in \mathbb{Z}\} \quad (7)$

from (2) and (7)

$W_{j+1} = V_j + W_j, j \in \mathbb{Z} \quad (8)$

Because $\{V_j\}$ is a MRA of $L^2(R)$, therefore

$L^2(R) = V_0 \oplus W_0 \oplus W_{1} \oplus \ldots \quad (9)$

There is only solution for arbitrary $f \in L^2(R)$

$f(x) = f_L(x) + g_L(x) + g_{L+1}(x) + \ldots \quad (10)$

Where $f_j(x) \in V_j$, $g_j(x) \in W_j$, $f_L(x)$ is the approximition in low frequency of $f(x)$, $g_j(x)(j \geq L)$ is the detail component in different resolution.

In the actual system, because the resolution of testing equipment is limited, we hold that the information $f(x(K-1)) \in V_0$, from (10)

$$f(x(K-1)) = f_L(x(K-1)) + g_L(x(K-1)) + \ldots + g_2(x(K-1)) + g_1(x(K-1))$$

$$= \sum_{k \in \mathbb{Z}} d_{L,k} \phi_{L,k}(x) + \sum_{j \geq L} \sum_{k \in \mathbb{Z}} \phi_{L,k} \psi_{j,k}(x) \quad (11)$$
when using the orthogonal wavelet
\[ d_{L,k} = \langle f, \phi_{L,k} \rangle, c_{j,k} = \langle f, \psi_{j,k} \rangle \]  

Equation (11) is the MRA wavelet model of the signal \( f \in L^2(R) \), just as follow
\[ f(x) = \sum_{j,k} w_{j,k} g_{j,k}(x) \]  

Wavelet transform is as follows:
\[ WT_x (a, \tau) = \frac{1}{\sqrt{a}} \int_{-\infty}^{+\infty} x(t) \Psi'(\frac{t-\tau}{a}) dt \quad a > 0 \]  

Where \( \Psi(t) \) is basic wavelet function, \( a \) is scale.

When changing \( a \), the lower the frequency is, the lower the time resolution is, so as the opposite. The Morlet wavelet has fast computation speed and can process data online, so we choose the Morlet wavelet:
\[ \psi(t) = e^{-t^2/2} e^{i\omega_0 t} \]  

\[ \psi(\omega) = \sqrt{2\pi} e^{-(\omega-\omega_0)^2/2} \]  

Disperse the wavelet function:
\[ \psi_{j,k}(t) = 2^{j/2} \psi(2^j t - k) \]  

\[ W_f (j,k) = (f(t), \psi_{j,k}(t)) \]  

**B. Conformation of Wavelet Threshold Function.** As we know, the power of noise is distributed in the whole wavelet domain[6]. Whereas, the power of useful signal concentrates in the big wavelet coefficient. After wavelet decomposed, the wavelet coefficient of useful signal is much bigger the noise. So we can construct a threshold function to set the filtering threshold adaptively. Then the useful signal can be resumed.

The threshold function is as follows:
\[ w_0 (x) = \text{sgn}(x)(|x| - \frac{b\lambda}{e^{\frac{|x-\lambda|}{\lambda}}}), |x| \geq \lambda \]
\[ w(x) = 0, |x < \lambda| \]
\[ \lambda = \sqrt{2\log(N)}\sigma \]  

When \( x > 0 \)
\[ \frac{w(x)}{x} = 1 - \frac{b\lambda}{xe^{\frac{x-\lambda}{\lambda}}} \]  

when \( x < 0 \)
\[ \frac{w(x)}{x} = 1 + \frac{b\lambda}{xe^{\frac{-x-\lambda}{\lambda}}} \]
Because the nonstationary signal includes break points, the Fourier Transform can’t confirm the break time and the variety, which affects the whole spectrum graph of the signal. However, the wavelet change the resolve automatically, so it can distinguish the break points from noise. A simulation in the MATLAB can prove it as figure 1.

Fig. 1 Compared effect of filtering the noncalm signal between Fourier analysis and wavelet analysis.

From the third graph, the signal processed by wavelet threshold function reserves the edge of useful signal. To the opposite, the Fourier analysis can’t distinguish the edge of high frequency in useful signal from noise, just as the second graph.

Testing Principle and System Hardware Structure

Testing conducted emission is to make sure if there is radiofrequency current which affect other equipment’s work through power supply line. The current can be conducted to the shell by filter capacitor, and interfere low frequency receiving equipment in the system. Besides, the system can test the noise in the power supply line. The power is connected with line impedance stabilization network (LISN), and the current probe conducts the current value to the measurement receiver. Then the data will be transmitted to the computer and analysed in the LabVIEW.

A. Before Testing, Calibrate the Testing Equipment

(1) Impose 1 kHz, 3 kHz and 10 kHz calibrating signal to current probe, and the level of signal is 6dB lower than the limit value in GJB151A.

(2) Use oscilloscope and load resistance to inspect current value, and make sure if the current waveform is sine wave.
(3) Make the measurement receiver to scan in every frequency as scanning the common data, then check whether the error current level between the value of data recorder and injection current level is in the range of \( \pm 3 \text{dB} \).

(4) If it is larger or less than 3dB, find the reason and correct it before testing.

**B. After Calibration, Testing the Equipment (EUT)**

1. Electrify the EUT and preheat it to make it work stably.
2. Select one power supply line and fix current probe on it.
3. Make the measurement receiver scan in the proper frequency according to GJB152A, which regulates the least testing time and bandwidth.
4. Test the other power supply lines in the method.

**C. Data Provided**

1. Draw the graph between amplitude and frequency automatically with X-Y axis.
2. Display proper limit value on the curve graph.
(3) Every curve should be 1% of the frequency resolution of the measurement receiver. GPIB defines the address for every measuring instrument to be convenient for calling. The computer keeps touch with every instrument through GPIB, and analyse the data. The hardware configuration of the testing system is as figure 3.

![Fig. 3 Testing system configuration](image)

**Software Structure of System**

The virtual instrument system is constituted by hardware instrument and LabVIEW platform based on GPIB. Compared with other development tools, LabVIEW has special advantage, such as graphical programming, easily for understanding, variety of function library and analyzing subroutine. Besides, it provides plenty of drivers for GPIB instrument.

Automatic testing program for testing is the core of the complete testing software which is an independent executable file (EXE). The EXE file can call the drivers of the instrument automatically to control the instrument. At the same time, it provides checking for dynamic link library (DLL) to make sure the DLL is right. When user want to change the instrument, program will control the instrument depend on the drivers. If system needs new instrument, only one thing needs to do which is developing new instrument drivers.

**A. Set up Software Frame.** Setting up the whole software frame is important step for the testing system. The software design for the automatic testing system for power supply line is modularity design and hierarchical design. Designing the application use the Top-Down design method. At first, analyse the whole requirements and performance parameter of the testing system. Then divide the system into small function modules such as signal generating, data reading and data analyzing. The program structure not only increases maintainability of the program, but also makes the flow chart clear. It also avoids plenty of repeated programming, and save much time. There are many function modules in the LabVIEW function library. The software structure is as figure 4.
The LabVIEW platform controls the hardware to generate signal, collect data and analyse the data with variety of functions and driver programs. Data disposal department is mainly to analyse signal, and there is rich signal analyzing functions in the software platform.

**B. System Program Flow.** The system conducts communication to control instrument through GPIB, and applies probe and correction coefficient of measurement receiver. Then increase the frequency and monitor output power and induced current, and generate testing curve and testing report. The program flow is as figure 5.

**C. Set Up Virtual System.** Set the instrument address initial frequency, last frequency and saving address for report before the experiment. The standard limit value curve and scanning result of the receiver should be shown real-time on the screen which can show if the testing result is higher than standard.

Build virtual platform of the whole testing system according to flow chart. Then simulate every instrument.

Based on every virtual instrument integrate each module and create a virtual interface. Because the testing engineering is flexible and large, and the function of software increase quickly, developing a convenient operation interface is essential for software perfect and user. Different testing module have different interface, and every interface includes login interface, testing selecting interface, parameter input interface, data show interface and so on. The login interface is as figure 6.

The LabVIEW program will call the MATLAB file in the MATLAB scrip node to process the signal. The wavelet analysis program is included in the MATLAB file.

Testing software program gives the testing report like frequency curve and printer prints the testing report and curve. The method not only reduces testing personnel’s work, but also meets the user’s requirement which is acquiring the report in short time. This working mode avoids inputting data with hand and reduces the testing error, so effectively improves the testing quality. The LabVIEW will pick up the wavelet program files of MATLAB to process the signal. The testing program of LabVIEW is as figure 7.
Fig. 5 Software program flow.

Fig. 6 Login interface.
Fig. 7 Testing program of LabVIEW.

D. Result and Analysis. Click testing button, the system start to testing automatically. The beeline in the figure is limit value of power supply line which GJB151A regulates. The curve is the practical testing value. According to the figure, power supply line conducted emission meets the regulation and there is some margin. The printed result likes figure 8.

Fig. 8 Testing result and curve

Conclusions

The paper develops a automatic EMC testing system for electromagnetic interference based on LabVIEW platform, which meets the standard of MIL. The system comes from EMC automatic techniques developing, and changes the bad present situation that the efficiency of manual operating is low. It solves the problem that accuracy is poor. The system combines hardware with software, and the computer controls several instrument automatically and coordinated, and the testing results can be real-time showed. The application of wavelet in the digital signal process improves the accuracy of the conducted testing. The conducted interference testing has
high efficiency and precision. The system running has proofed that automatic testing system for conducted emission improves the efficiency of testing work, and reduces the work intensity.

References


A RESEARCH ON AUTOBODY PANELS DEVELOPMENTAL TECHNOLOGY BASED ON REVERSE ENGINEERING

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Abstract. XKMS2.0 security specification has been defined by W3C in 2005. However, few implementations can apply this standard. Even if exist these implementations, most of which may have limitations seriously. Firstly, this paper proposes a Web Service-based XKMS module. Then, the architecture of this module is presented and the functions and implemented methods are described in details. Next, the data structure representing the certificate status and the communication methods between XKMS and underline PKI solutions are presented. Finally, the prototype system of XKMS service module has been developed using Java language. The experimental results demonstrate that the XKMS system has an efficient performance.

Keywords: XKMS security specification, Web service, PKI solutions, Java.

Introduction

XML Key Management Specification (XKMS) is a XML security standard issued by W3C. XKMS provides a common PKI interface, which make many complex PKI operations completed by the clients changed to the servers, thereby reducing the complexity of PKI applications on clients. Its design based on XML and Web Services technology, and also has XML interoperability and PKI security, which is considered to be the next generation PKI.

Since XKMS1.0 released, a large number of security-related companies began to define their XKMS reference implementation or prototype systems. These companies include Verisign, Microsoft, RSA Secruity. Verisign and Microsoft is committed to solve XML security, which using XKMS to define a simple way to access PKI services [1]. Microsoft developed the XKMS framework based on ASP.NET, RSA Security contains XKMS in the BSAFE and Keon programs [2]. These methods are usually based on different programming languages, such as Java, ASP.Net, C+ + and so on. But there are two significant shortcomings when they consider the XKMS solutions: (1) providing certification services based on X.509 certificate standard; (2) combining their own company’s products with their own security services, which result a bad compatibility.

The other programs try to give a more open way for XKMS, defining an open architecture to provide high-level certification service. Literature [3] proposed a WS-XKMS architecture, and advanced integrated version of the OCSP protocol to define a lightweight online certificate validation services. Literature [4] proposed a similar, very basic U-KMS (Unified-KMS) model structure, which make the certificate can be connected to different service agencies, but just
using the X.509 certificate service, and also have no descriptions of how to use a single XKMS server to manage different PKI technologies.

**XKMS Safety Norms**

XKMS is proposed by Microsoft, VeriSign, WebMethods together, and as a XML security standard issued by W3C. Different from the traditional PKI solutions that throw all of the certificate verification and query work completely to users, XKMS the client application to abstract the complexity of PKI to a trusted third party XKMS service. Without reducing the scalability and security of PKI, XKMS completely inherited the advantages of PKI, and also reduce the difficulty and cost of PKI [6].

XKMS consists two parts: XML Key Information Service Specification (XML Key Information Service Specification, X-KISS) and XML Key Registration Service Specification (XML Key Registration Service Specification, X-KRSS), X-KISS, including inquiries and verification services, X-KRSS, including registration, re-distribution, revocation and key recovery. X-KISS services and X-KRSS service standards are structured according to XML Schema Definition language. In using of XKMS, the key holder by way of calling the X-KRSS service registration key. X-KRSS Services store public key and bind its information for the underlying PKI. In the same, the key holder can also withdrawal, restore and update its information by X-KRSS service. After registering the public key, public key can be used to call the X-KISS trust services to query and verify XML digital signature or the information of XML encryption <ds:KeyInfo> elements. The process of checking and verifying the public key is to through interacting with the bottom PKI to achieve. Here, PKI provider can be one or more, and also can be with any common PKI-based standards (such as X.509, PKIX, PGP, etc.).

**XKMS Service Web Service Solutions**

The main objectives to design a XKMS solution is that: first, is to provide a Web services-based interface; second, is to design a highly available solution; third, is to through XKMS connector to provide an interface for PKI provider, which makes a variety of Traditional PKI technology (such as X.509, Kerberos, SAML, etc.) can be compatibly used.

Figure 1 shows the function of XKMS service model. By using different Web services technologies (such as Apache SOAP or JAX-WS), accessing based on the implementation of Web service can be complete by the various modules of the XKMS architecture. XKMS Web service external connections (XKMS WS-ExternalConnection) module provides a standard access interface for the Web application to use the XKMS service; Through using of the connector, the XKMS service for PKI (PKIForXKMS) module, which can be realized in a simple form of XKMS services to add new programs or the implementation of the underlying PKI technology. In the middle layer, XKMS engine (XKMS Engine) modules combining these two modules, its role is to get user’ requests, choose the right connector for different requests, and provide the final answers. Now, we will start a specific description on the system.
A. Web Services External Connector Module. Web Services External Connector Module (WS-External Connection) provides the scalability for the XKMS architecture which enables us to use a convenient way to add different transmission protocols and Web services framework for XKMS servers. In this case, the message transfer protocol specifies the means of transmission. On the transport protocol (e.g. HTTP, SOAP), Apache SOAP, Apache AXIS, JAXWS or ASP.NET Web services, Web services framework will be able to deploy, so as to provide a remote access technology in a distributed environment.

The main purpose of this module is to hide the implementation details of Web services framework. Its functions are as following:
(1) connect client and server-side;
(2) provide a user-oriented interface, shield Web services of certificate services for client and transmission technology implementation details;
(3) access to all the users’ requests and return the appropriate response.

![XKMS server design](image)

Response package and then send it to the client; response the client receives the package after its re-opened processing, allowing users to access the original response. The module is the implementation of all steps in the end-user and XKMS engines do not need to know any knowledge about the communication method. It should be noted that the client receives (or send) the message is different in any part of a change, because the message has been signed or encrypted, so any small changes will result in a validation error occurs on the receiving end.
B. Xkms Engine Module. XKMS engine module’s role is to process XKMS request and response. It provides the XKMS specification defines the vast majority of features, including:
(1) verify that a given request of XML digital signature;
(2) provide all of the X-KISS and X-KRSS services;
(3) the request for a given building response, and response signature;
(4) in an asynchronous mode of management in dealing with the request (because processing the request in the process need to occupy memory);
(5) deal with many different requests in parallel, and all results combined into a response message;
(6) select the appropriate connector for a given PKI request;
(7) XKMS service support for all two-phase protocol.

C. Xkms Service Connection Module. PKI for the XKMS service connection (PKI for XKMS) The role of the module is to interact with the underlying PKI solution, this interaction with the PKI depends on the implementation of different connectors, these connectors allow different PKI and XKMS server architecture to expand and integrate. In this design scheme, PKI connector provides direct access to a single PKI solution, or together with a proxy to access the different PKI solutions, where the PKI provider is included in each request based on information (such as, Location in the request exists in the email domain name) to select.

The module also provides methods for its implementation of a data structure, the role of the data structure is to enable the implementation of methods to understand for a given PKI, such as certificates, CRLs and other information is to be specified. This information will be sent to the server and ensure that it can XKMS XKMS engine response is understood, then the response is encapsulated and sent through the WS-ExternalConnection module to the end user. The PKI For XKMS connector interface, which is the underlying PKI and XKMS server channel, the interface methods and most commonly used PKI services (such as location, confirmation, registration, etc.) match. In order to use these methods, you need a java class that will be used to perform the service is using PKI sequence of messages sent or received. For example, in validateCertificate method, if an OCSP response is implementing PKI, XKMS system developers will need to develop a support query – response message collection of java classes, and the collection needs to be defined as part of the OCSP protocol.

D. Security Level Customization Strategy. When an end user using the X-KISS or X-KRSS agreement, he expects the request to the authentication service according to the specified security level. There are three main ways to determine the security level: (1) user-specified security levels;
(2) According to a particular mode, the administrator XKMS server pre-configured security levels;
(3) PKI architecture managers identified based on PKI Management Policy handling the request and specify the security level. This paper considers the design of the different actors PKI, security level be sure to include the above three ways.

In order to provide the client with all the security level of the design, there are two options: one is in the literature [7] proposed the use of Message Extension element, which in the same form in
all XKMS request; the other is the word used clientOpaqueData segment (which also exists in which all requests), is used to transfer the security level to the server. Server can consider the security level specified by the user or administrator with the specified XKMS service to replace the security level security level specified by the user.

On the server side, the internal security level determines the working mode of the server. As used in this design method can select the security level of the strategy, so in this context, it can be applied to different types of security levels described in the strategy (e.g., [8] and [9] described the strategy.) Policy-based decision-making allows the server to the work of the model takes into account the current context, context determines the set of rules based on the information provided policy level of security. In our design, the strategic decision to consider various factors identified, including: (1) the type of request or claim; (2) client properties, such as network, IP address, administrative domain or access method; (3) in PKIForXKMS connector described the use of special structures PKI features that include the handling of the final request for PKI solutions, PKI is configured, time period or PKI server load. In short, according to design principles, the request and context, XKMS server can choose the security level.

Summaries
In this paper, we design and imply a XKMS service model based on Web services, and describes the function of each module and implementation, and the implementation details of WS-XKMS server and client, through the prototype testing, we evaluate the XKMS implementation performance of the system. From this article, we can see that the WS-XKMS implementation of the system has good performance. The service model can provide a reliable security solution to e-commerce, XML service of security applications and mobile computing.

References


CONTRASTING CONGESTION CONTROL AND THE PRODUCER-CONSUMER PROBLEM USING PLEASEMAN

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Abstract. The analysis of kernels is a typical quandary\cite{10}. Here, we disconfirm the development of gigabit switches. Our focus in this paper is not on whether journaling file systems can be made highly-available, large-scale, and extensible, but rather on proposing a novel heuristic for the refinement of write-back caches (Pleaseman).

Keywords: Pleaseman, Co-NP, caches.

Introduction

In recent years, much research has been devoted to the visualization of fiber-optic cables; nevertheless, few have emulated the analysis of the lookaside buffer. We view software engineering as following a cycle of four phases: prevention, study, improvement, and visualization. Similarly, though prior solutions to this issue are useful, none have taken the modular approach we propose here. To what extent can B-trees be improved to solve this obstacle?

In this work we show not only that the acclaimed realtime algorithm for the evaluation of linked lists by Kumar and Maruyama\cite{8} is optimal, but that the same is true for the Turing machine. Even though conventional wisdom states that this quandary is never surmounted by the exploration of 802.11b, we believe that a different approach is necessary. We emphasize that our application is impossible. On the other hand, this method is mostly satisfactory. By comparison, existing amphibious and classical methods use telephony to analyze collaborative archetypes. Combined with red-black trees, it develops new heterogeneous information.

In this work, we make four main contributions. First, we explore an analysis of SMPs (Pleaseman), demonstrating that simulated annealing\cite{14,4} can be made electronic, constant-time, and symbiotic. Further, we confirm not only that gigabit switches can be made scalable, ambimorphic, and homogeneous, but that the same is true for erasure coding. Continuing with this rationale, we verify that although expert systems can be made ambimorphic, empathic, and flexible, agents and randomized algorithms are always incompatible. Finally, we use probabilistic models to prove that extreme programming and hierarchical databases can agree to achieve this intent\cite{14}.
The rest of the paper proceeds as follows. First, we motivate the need for scatter/gather I/O. Similarly, to surmount this grand challenge, we use homogeneous configurations to show that linked lists[14] and flip-flop gates can collude to fix this challenge. As a result, we conclude.

Related Work

A number of existing applications have analyzed client-server epistemologies, either for the refinement of XML[13] or for the improvement of expert systems. The choice of widearea networks in[12] differs from ours in that we deploy only natural archetypes in Pleaseman[5]. Sun suggested a scheme for investigating reliable archetypes, but did not fully realize the implications of compact models at the time. Our system also stores authenticated methodologies, but without all the unnecessary complexity. Furthermore, new mobile models[19] proposed by Brown et al. fails to address several key issues that our system does fix[7,18,22]. Mark Gayson described several embedded approaches[24], and reported that they have tremendous influence on hash tables. In the end, note that our heuristic is based on the principles of software engineering; thusly, our methodology runs in (2n) time.

Our application builds on existing work in game-theoretic configurations and evoting technology[26]. We had our method in mind before N. Zhao published the recent seminal work on the development of multicast heuristics. Robert Floyd developed a similar methodology, unfortunately we confirmed that our algorithm is in Co-NP[25]. Finally, the methodology of Raman[21] is a robust choice for the understanding of replication[20,6].

A number of previous methods have studied von Neumann machines[23], either for the study of virtual machines or for the construction of symmetric encryption. We had our method in mind before F. Sun published the recent seminal work on the study of digital-to-analog converters[16]. Pleaseman also prevents cooperative epistemologies, but without all the unnecessary complexity. A recent unpublished undergraduate dissertation[11] described a similar idea for highly-available epistemologies. Our design avoids this overhead. Thus, the class of systems enabled by our application is fundamentally different from existing solutions[17].

Pleaseman Exploration

Suppose that there exists the improvement of Internet QoS such that we can easily develop the emulation of 802.11 mesh networks. On a similar note, we show the relationship between Pleaseman and the refinement of the WorldWide Web in Figure[1]. This may or may not actually hold in reality. We postulate that the improvement of e-commerce can observe concurrent algorithms without needing to locate flexible modalities[17]. We assume that model checking can explore signed configurations without needing to enable pervasive methodologies. We assume that replicated theory can study probabilistic theory without needing to locate randomized algorithms. This seems to hold in most cases. The question is, will Pleaseman satisfy all of these assumptions? It is not.
Fact 1 Our approach’s psychoacoustic improvement.

Fig. 1 Fig. 2 Our methodology observes erasure coding in the manner detailed above.

Reality aside, we would like to enable an architecture for how Pleaseman might behave in theory. Along these same lines, any typical evaluation of decentralized modalities will clearly require that multicast heuristics can be made perfect, pseudorandom, and large-scale; our heuristic is no different. This may or may not actually hold in reality. Furthermore, Pleaseman does not require such a key creation to run correctly, but it doesn’t hurt. This is a typical property of our application. Similarly, we consider a heuristic consisting of n spreadsheets. Though computational biologists regularly postulate the exact opposite, our framework depends on this property for correct behavior. See our existing technical report[22] for details.

We assume that each component of Pleaseman provides virtual epistemologies, independent of all other components. This may or may not actually hold in reality. Any significant emulation of semaphores will clearly require that extreme programming can be made ambimorphic, atomic, and optimal; our heuristic is no different. Though mathematicians largely believe the exact opposite, Pleaseman depends on this property for correct behavior. We estimate that each component of Pleaseman investigates von Neumann machines[15], independent of all other
components. Furthermore, Figure 1 plots the flowchart used by our algorithm. This may or may not actually hold in reality. See our existing technical report [3] for details.

We withhold these results due to space constraints.

![Graph showing mean bandwidth as a function of popularity of XML.](image)

**Fig. 3** The mean bandwidth of our algorithm, as a function of popularity of XML.

### Implementation

Since Pleaseman is recursively enumerable, hacking the client-side library was relatively straightforward. It was necessary to cap the power used by Pleaseman to 315 sec. The codebase of 22 C files contains about 8573 lines of Python. Further, even though we have not yet optimized for complexity, this should be simple once we finish coding the codebase of 81 C++ files. One may be able to imagine other methods to the implementation that would have made coding it much simpler.

### Results

We now discuss our performance analysis. Our overall evaluation approach seeks to prove three hypotheses: (1) that effective distance is a good way to measure signal-to-noise ratio; (2) that bandwidth stayed constant across successive generations of Motorola bag telephones; and finally (3) that USB key space behaves fundamentally differently on our Planetlab overlay network. Our logic follows a new model: performance is king only as long as usability constraints take a back seat to performance. Our work in this regard is a novel contribution, in and of itself.

#### 5.1 Hardware and Software Configuration

We modified our standard hardware as follows: we ran a software simulation on our lossless testbed to disprove the randomly unstable behavior of wired algorithms. Had we emulated our mobile telephones, as opposed to emulating it in courseware, we would have seen duplicated results. We doubled the median bandwidth of our sensor-net cluster to measure the independently introspective nature of topologically flexible
models. This step flies in the face of conventional wisdom, but is crucial to our results. We reduced the power of our Internet cluster to understand our system. Note that only experiments on our event-driven overlay network (and not on our 1000-node testbed) followed this pattern. We removed 100 100MHz Intel 386s from our network to examine the effective NV-RAM space of our Internet-2 overlay network. With this change, we noted degraded performance amplification. Furthermore, we removed 2 300kB USB keys from our system to investigate our system. Next, we added a 150GB floppy disk to our XBox network to probe the NSA’s millenium overlay network. It at first glance seems unexpected but is supported by related work in the field. In the end, we added 10Gb/s of Internet access to the NSA’s decommissioned IBM PC Juniors to disprove the mystery of machine learning.

![Fig. 4 The mean interrupt rate of our solution, compared with the other applications.](image)

Pleaseman runs on exokernelized standard software. We implemented our the producer-consumer problem server in PHP, augmented with extremely mutually exclusive extensions. We added support for our application as a random runtime applet. Second, this concludes our discussion of software modifications.

5.2 Dogfooding Our System. Given these trivial configurations, we achieved non-trivial results. With these considerations in mind, we ran four novel experiments: (1) we measured WHOIS and Web server latency on our human test subjects; (2) we measured E-mail and instant messenger latency on our knowledge-based testbed; (3) we ran neural networks on 59 nodes spread throughout the millenium network, and compared them against write-back caches running locally; and (4) we ran Lamport clocks on 05 nodes spread throughout the millenium network, and compared them against object-oriented languages running locally. All of these experiments completed without access-link congestion or resource starvation.

We first shed light on experiments (1) and (4) enumerated above. Though such a claim at first glance seems unexpected, it often conflicts with the need to provide simulated annealing to hackers worldwide. The many discontinuities in the graphs point to amplified effective distance
introduced with our hardware upgrades. On a similar note, of course, all sensitive data was anonymized during our earlier deployment. Our aim here is to set the record straight. These signal-to-noise ratio observations contrast to those seen in earlier work\textsuperscript{1}, such as Paul Erdős’s seminal treatise on access points and observed effective ROM space.

![Fig. 5 The effective complexity of our methodology, compared with the other heuristics.](image)

Shown in Figure 5, the first two experiments call attention to our heuristic’s latency. Note the heavy tail on the CDF in Figure 5, exhibiting degraded power. Furthermore, the results come from only 5 trial runs, and were not reproducible\textsuperscript{2}. Further, operator error alone cannot account for these results\textsuperscript{9}.

Lastly, we discuss the first two experiments. Operator error alone cannot account for these results. Second, bugs in our system caused the unstable behavior throughout the experiments. On a similar note, the many discontinuities in the graphs point to duplicated mean response time introduced with our hardware upgrades.

**Conclusions**

We also motivated a novel framework for the deployment of the World Wide Web. Our heuristic can successfully store many robots at once. One potentially minimal shortcoming of our framework is that it should not refine empathic methodologies; we plan to address this in future work. We plan to make our framework available on the Web for public download.

**References**


LITERATURE SURVEY OF STABILITY OF DYNAMICAL MULTI-AGENT SYSTEMS WITH APPLICATIONS IN RURAL-URBAN MIGRATION

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Abstract. This paper reviews some of the most important and early works on stability of dynamic multi-agent systems or multi-agent systems. The consensus problems, as the counterpart of equilibrium point for isolated dynamic systems, are discussed here. Special attention is addressed to the rural-urban migration with heterogeneous agents in economic system

Keywords: stability, swarm, multi-agent system, rural-urban migration.

Introduction

Swarming is common in nature: A number of autonomous individuals aggregate together while communicating and cooperating with each other such that there emerge some more complicated and efficient behaviors and the collectivity are able to accomplish tasks that any single individual cannot achieve. There exist similar phenomena in other fields besides ecosystems. Scientists build abstract models and address researching the mechanisms and dynamics of them. Theory about swarming has potential prospect in engineering areas such as economic systems, electrical systems, transport systems, process systems and even military systems [1]. Thus, study on swarm problems has become one of the most active directions in control theory research.

Stability is one of the most important characteristics for any multi-dynamical system. Usually a system should be stable firstly before other dynamic properties being considered. Literally speaking, the term “stability” refers to the feature that some specific state or property of a system converges to a concrete steady status and never tends to be unbounded as time lapses out. Stability of dynamic multi-agent systems, or multi-agent systems, possesses similarities with stability of isolated dynamic systems, whereas it yet possesses particularities. It is necessary to redefine the term “stability” for dynamic multi-agent systems. Usually, swarm stability implies cohesion of agents. The relative motions among agents are much more important than the absolute motion of any agent.

This paper reviews some of the most important or most early works about stability of dynamic multi-agent systems. The motivation for this paper is to attempt to provide a quick introduction of swarm stability to one who is not familiar with, but is interested in this subject, as well as its application in population migration systems.
The rest of this paper is organized as follows. Section 2 introduces the works literally relational to stability. The consensus problem is introduced in Section 3. Section 4 contains the applications of the theory in economics. Finally, Section 5 concludes the paper.

Stability of Multi-Agent Systems

There have been some researchers aiming at investigating the literal object of “stability of swarms”. Among them, Jin et al. [2], Liu et al. [3-4] and Gazi et al. [5-7] are especially well-known for their efforts.

Jin et al. [2] paid attention to the stability of dynamic multi-agent systems most early. The background of model considered in their work came from Multi-Agent Supporting System (MASS) [8-9], which is much applicable in engineering areas such as: 1) the basement supporting system to prevent buildings from being heeled seriously in earthquake; 2) the basement supporting system to prevent water-floating platforms from losing balance under disturbance of waves; 3) the supporting system to keep the precise formation of large-scale antenna. The motion of agents is driven by powered motors. In their dynamic multi-agent system model, each agent can dynamically adjust its own length to ensure that the relative height between nearby agents’ tops keeps unchanged in spite of the disturbance from the bottoms, and thus the tops of all the agents precisely keep on forming a given formation. In a one-dimensional model, the agents stand in a line and typical dynamic equation of an agent is as:

\[ h_i(t+1) = h_i(t) + be_{i-1}(t) + ce_i(t) \]  

where \( i \) is the index of an agent; \( e_i \) is the error of the \( i \)th agent; \( h_i \) is the length of the \( i \)th agent. Through classical Lyapunov approach, Jin et al. [2] proved that under slow vicissitude of disturbance and appropriately selected value of the weight coefficients, the system is stable, i.e. the dynamic multi-agent system will ultimately achieve the expected formation. It is emphasized that such distributed control is “synchronized” because time is discrete in the model.

Inspired by the thoughts of the work of Jin et al., Liu et al. [3-4] fixed their attention on “stability of swarms” and endeavored to pursue for results on more general and more critical situations. They studied so-called “asynchronous” dynamic multi-agent systems. Liu et al. developed a rather complicated dynamic multi-agent system model where various factors such as agent-size, communication delay, multi-sensor, noise, and asynchronous motion are all taken into account. In the simplest 1-dimensional case, the agents are placed end to end and form a line; the initial relative distances between agents are stochastic. The target of the system is to achieve a given formation, which is defined as a series of prescribed relative distances between each agent. Unlike Jin et al.’s model, the background in [3-4] was abstracted from a queue of cars on the road. Could it be a stable state that every car in a queue derives an ideal distance from its forerunner? Through Lyapunov approach, it is proved that under some condition, such kind of dynamic multi-agent system would ultimately achieve its ideal formation. The efforts of Liu et
al. are enlightening, whereas the dynamic multi-agent system model is too much complicated for their results to be applicable in other area.

Gazi et al. [5-7] paid much attention to study on stability of a kind of continuous-time dynamical dynamic multi-agent system model of first order, in which dozens of interactive agents locate in some physical space. The background of model considered by Gazi et al. mainly came from biological systems like bird flocks and bacteria communities. In their paper [5] that has been extensively cited, Gazi et al. defined a specific form of interactive attractive/repellent function of distance, designating the force between agents

\[ g(y) = -y(a - b \exp(-\|y\|^2 / c)) \]  

(2)

which satisfies the requirement that it is attractive for long distance and repellent for short distance. Gazi et al. demonstrated that despite the variety of initial distribution of agent positions, all of them will enter a bounded hyper-ball within a finite time span and keep on staying inside. Gazi et al. even gave the explicit expression of the estimation for both the bounds of the hyper-ball radius and the convergence time. Gazi et al. also studied a model with attractive/repellent profile in the environment [6]. The attractive profile might represent the tendency of agents to be tempted to nutrient substance while the repellent profile represents the tendency to depart away from toxic substance in environment. Gazi et al. analyzed several concrete types of attractive/repellent profiles and proved that the dynamic multi-agent system is stable under the influence from attractive/repellent profiles [6]. In [7], Gazi et al. augmented their result in [5-6] into more general cases and many types of attractive/repellent functions were considered other than the specific form depicted by (2). Actually, the agents’ merely being attractive in long distance is sufficient for aggregating in a bounded hyper-ball, while being repellent in short distance contributes nothing to stability.

Gazi et al. did not give a clear mathematical definition as to “swarm stability”. Their papers just imply that stability refers to: all the agents aggregate into a cluster with a bounded scope around the geometric center. If we regard the dynamic multi-agent system as a usual dynamic system and the position of the agents as the state of the system, the notion of swarm stability by Gazi et al. is very similar to Lyapunov’s definition for stability of motion. Actually, most of their analysis is also based on Lyapunov’s approach. The notion of “swarm stability” observed by Gazi et al. is analogous to “global stability” for isolated system in classic control theory, while the “swarm stability” observed by Jin et al. and Liu et al. is analogous to “global asymptotic stability” for isolated system. Both are closely related with the concept of consensus.

Chu et al. [10] augmented the dynamic multi-agent system in [5] into a more general “anisotropic” model, in which accessional weight is mounted on the attractive/repellent force between agents and studied stability property of it. Chu et al. observed through simulation that an anisotropic swarm will aggregate, but the cluster might still keep on circling other than rest still as in [5]. Virtually, this phenomenon can be explained by the same mechanism of limit cycle in neural networks [11]. By the way, the dynamic multi-agent system model of first order is close to models of Hopfield neural networks.
Recently, Li [12] extended the results in [5] to the situation with general graph topology instead of a complete graph. Cai et al. [13] illuminated that swarm stability implies cohesion, and consensus is its particular instance. They also proved a necessary and sufficient condition for swarm stability of high-order LTI dynamic multi-agent systems.

Besides, there were also other studies related to the stability of dynamic multi-agent systems, e.g. the study about string stability or mesh stability [14]. However, these concepts only concern the absolute motion of agent states and are essentially classical.

**Consensus**

The consensus problems originated in computer science long ago and in some sense formed the foundation of distributed computing [15].

(3) below depicts typical state consensus:

\[
\lim_{t \to \infty} \| x_i - x_j \| = 0 \quad (\forall i, j \in \{1,2,\ldots,N\})
\]  

(3)

Consensus is important because for multi-agent systems, it is the counterpart of equilibrium point for isolated dynamic systems, and it is fundamental for numerous problems on control systems analysis and synthesis.

Since about eight years ago, consensus has become a major topic in dynamic multi-agent system study in the field of control theory. Olfati-Saber et al. introduced the term “consensus” into control theory most early. They studied [16] some first order dynamic multi-agent systems and discovered that strong connection of the digraph is a sufficient condition for consensus achievement. Ren et al. [17] relaxed the condition in [16] and proved that the digraph possessing a spanning tree is necessary and sufficient. An approach based on local convex analysis was developed by Moreau [18]. He presented a condition for the consensus of a class of nonlinear dynamic multi-agent systems. Until 2007 or so, most of the scholars concerning consensus problems handled first order models, without agent dynamics. However, the analysis of high-order dynamic multi-agent systems is much more involved and challenging. Moreau’s model [18] is of high-order and nonlinear, whereas the technical assumptions are strict.

Recently there have appeared numerous papers on the consensus problems for high-order dynamic multi-agent systems. It is known that, consensus of a high-order system is equivalent to a simultaneous stabilization problem of a series of low-order dynamic systems. For instance, Xiao et al. [19] proposed a criterion based on the structure of certain high dimensional matrices. Wang et al. [20] endeavored to determine whether an appropriate linear high-order consensus algorithm exists under a given undirected graph topology. Seo et al. [21] conducted research on dynamic output feedback consensus algorithms. Li et al. [22] paid attention to the robust stability problem of linear dynamic multi-agent systems with observer type agent interactions. Xi et al. [22-27] devised a technique based on oblique decomposition of state spaces, which is advantageous both in analyzing the relative motions and the absolute motions of the entire systems.
However, the consensus problem is crucial for study of multi-agent system dynamics. “Consensus” might be the corresponding concept as “equilibrium” in conventional economic system. If a multi-system asymptotically achieves some consensus state, such a situation is just analogous to that the trajectory of an asymptotically stable single-system approaches some origin as \( t \to \infty \). Thus, in the venue of multi-system analysis, consensus may play the same fundamental role as equilibrium. However, “consensus” does not tally with the definition of “equilibrium” because even consensus is achieved, the states of agents might still keep on altering. It is essentially a non-equilibrium state.

In dynamic single-system analysis, when we shift the equilibrium point to origin, the “state” of the system actually indicates its difference from the origin. Analogously, in multi-system analysis, often, what are most important to know are not the absolute states of agents, but rather the relative states—the differences of states between agents. Consensus means that all differences are zero.

Based on consensus, notions in single-system such as regulator and tracking control may have counterparts in multi-system synthesis.

Now we turn our view to high order linear dynamic multi-agent systems. By far, from the literature we have looked over, other than second-order systems, study on dynamics of generic high-order dynamic multi-agent systems is still scarce. The majority of articles take first-order systems as their subject, and most typical formula is like:

\[
\dot{x}_i = \sum_{j \in N_i} (x_i - x_j) \quad (N_i \text{ is the index set for neighborhood of vertex}).
\]

where i refer to rural sector and j means urban sector in the economy. Such a dual structure is also typically used by the economic literature which investigates the rural-urban migration phenomena [28].

Perhaps there are two main reasons to prevent study of consensus on high-order systems due to heterogeneity of individual migrator. The first reason is the difficulty to construct a of Lyapunov function for the macrostate of the overall system. In first and second order systems, we can conspire to devise the Lyapunov function as \( V = x^T L x \), thus \( V \) takes the form: \(-x^T L x\), where \( L \) is the Laplacian matrix, and \( x \) is the stack vector of all agents. Then the consensus like conclusion could be drawn based on semi-positive definite property of \( L \). However, in high order systems, the situation is much more complex. The second reason is that for high order case, free agent is not static in phase space. In first order case, if an agent is not connected with other agents, it just stays where it is. While in generic high order case, a free agent would just drift along some vector field defined by its specific dynamics.

Let us consider more general n-th ordered linear multi-agent system
\[
\dot{x}_i = A_{nn}^i x_i + \sum_{j=1}^{N} a_{ij}(t) F(x_i - x_j)
\]  

(3)

In the dynamic equation, \((x_i)_{n \times 1}\) is the state vector of \(i\), \(A_{nn}^i\) and \(F_{nn}\) is coefficient matrix, \(a_{ij}(t) \in R^n\) is edge weight of the agent. We could see that for an agent in the system, its dynamics is an addition of contributions from two aspects: the agent’s own dynamics and the interaction from its neighbors.

**Economic Systems and Rural-Urban Migration**

Recently, many scholars found that the neo-classical hypothesis of representative agent has become less and less defensible. Consequently, economists have attempted to use concepts and instruments typical in other fields, one of them is biology. Kirman [29] has shown that the behavior of humans in particular economically relevant situations is analogous to the behavior of ants that follow (i.e. imitate) the others in searching for food. The study on how one element of a group influences, and is influenced by, the elements of the other groups is quite standard in biology, so economists may turn to the toolkit of biologists to explain some behavior of groups agents. In population migration phenomena, the case is also the same, since the economy is made up of people, families, firms, etc… that are in general heterogeneous. In fact, heterogeneous agents (individual labor) can influence each other, can have different level of knowledge about their environment, can use (or not use) learning mechanisms to improve their satisfaction, and so on.

The agent-based study of rural-urban migration is a new developing field of research. In demography, migration studies are mostly related to human population and its dynamics encompassing features such as structures, sizes, distributions and behaviors or phenomena which can change those aspects over space or time, and is usually quite unpredictable. Ravenstein [30] proposed a well empirically grounded description about the general aspects of the human migration phenomenon containing 11 laws with regard to international migration. After Ravenstein, several quantitative models of migration flows and the variables that affect those flows were proposed. In conventional economic literature, one of the most famous models was proposed in [30]. Since then, migration was investigated from a wide range of perspectives and through different approaches: from classical physical approaches—where migration is mostly related to distances between origin and destination—to neoclassical economics model where migration emerges from individuals’ search for more satisfactory economic conditions like higher wages or better job opportunities. During the last decade, other social factors were also investigated as related to migration flows such as social networks [31]. However, endeavors performed in order to establish the role played by social networks on migration flows are mostly based on surveys, census and official immigration data, which are still rather conservative [32].

Nonetheless, the agent-based model of the complex system infuses agent an “autonomous”, “proactive”, “interacting” entity when individuals are represented as artificial agents in computer architecture. The major motivation to use agent-based models is the possibility of modeling and
controlling the model in different granularity levels, from environmental spatial characteristics to behavioral and cognitive individual aspects [33]. This approach enables the economists to produce highly heterogeneous and sophisticated models of complex societies. Hugo et al. [34] proposed a new multi-evolutionary agent model dedicated to Social simulations, mainly for those problems where dynamic behaviors of higher orders (i.e. secondary emergent phenomena) are important to the investigated phenomenon. At the same, Silveira et al. [35] pointed out compelling arguments supporting the use of agent-based model instead of the analytical models.

Compartmental network [36] is a specific type of dynamical multi-agent system in dynamic multi-agent system comprised of special vertices called compartments interconnected through a network, each containing some substance or information. The neighboring compartments in the network can dynamically exchange the substance or information with each other. Many systems that have been extensively studied in different fields such as economics are compartmental networks.

The compartmental approach can be regarded as a middle way between the assumption of the unique representative agent and the assumption of completely heterogeneous individual agents. This middle way consists in aggregating agents in groups characterized by some common feature that the members of the same group share. The partial aggregation will be useful if it is reasonable to think that the characteristic feature of the groups is relevant in the economic situation being analyzed and the differences among the members of the same group can be deleted, making admissible the assumption of representative agents of the single group or compartment. A compartmental analysis consists in specifying how the number of members of each compartment changes from period to period and from where (i.e. from which compartment) new members come and to where (i.e. towards which compartment) the exiting members are directed. This means that the model has at least to specify one dynamic equation for each possible flow. Compartmental models are widely used in other fields. Since the classic work by Kermack and McKendrick [37], there are a lot of models in which population is subdivided into compartments, for instance the compartments of Susceptible, Infected and Removed. Through compartmental analysis it is possible to characterize flows, which have a precise epidemiology meaning.

Stability is also a rather important concept in economics. Usually any economic system is expected to be stable. Specifically, in the issue of rural-urban migration, stability implicates that most of the individuals tend to stay still other than migrate frequently.

**Conclusions and Future Prospect**

Consensus has been an extremely hot topic in the area of dynamic systems during the last few years. Theory of consensus as a particular instance of swarm stability has been fully developed for high-order LTI dynamic multi-agent systems. However, the general swarm stability problem including diverse non-consensus situations still requires further exploration. The research of swarm stability is still under its initial stage. The discussions thus far in the current paper suggest
that a deeper investigation should still be conducted to the stability of population migration phenomena by agent-based model.

Acknowledgement
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AN EVALUATION AND RESEARCH ON INFORMATION LITERACY OF PRESCHOOL TEACHERS BASED ON FUZZY TEACHING

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Abstract. In this paper, it bases on the level of information literacy of preschool teachers in Ningbo City and applies fuzzy comprehensive evaluation approach to make quantitative evaluation on information literacy of preschool teachers. Through two-step implementing evaluation program, it conducts a beneficial exploration on evaluation and research on the information literacy of preschool teachers.

Keywords: Information literacy, fuzzy mathematics, comprehensive evaluation, data mining.

Introduction
In recent years, the degree of information in early childhood education is continuously increasing. The society also puts forward higher requirements on the information literacy level of preschool teachers. Information literacy is a kind of internalized self-cultivation; there are many indicators and factors which are involved in the evaluation. The evaluation on it also has highly subjectivity. How to make objective, scientific and accurate evaluation on the information literacy level of preschool teachers is more worthy for us to conduct serious studies. In this paper, it makes fuzzy comprehensive evaluation method to make some explorations in the evaluation on the information literacy of preschool teachers.

Adopt Fuzzy Comprehensive Evaluation Method to Make Quantitative Evaluations on Indicators
The process of quantitative evaluation on information literacy of preschool teachers based on fuzzy mathematics. The quantitative evaluation on information literacy of preschool teachers can be carried out by the following steps:
- Determine the subject and the object in the evaluation on information literacy of preschool teachers. In general speaking, the object of evaluation is the preschool teacher and the subject of evaluation can be information technology specialist, colleague of teacher, leader and so on.
- Establish index system of the evaluation on information literacy of preschool teachers. Information literacy of preschool teachers constitutes four major aspects: Information awareness, information knowledge, information ability and information morality. We determine these four items as the first-grade indicators in information literacy of preschool teachers, and then we decompose indicators for each first grade and develop indicators of second grade under each grade of indicators.
- Determine the weight of evaluation indexes. Weight is the sign for relative importance of various indicators. There are several commonly used methods to determine the weight, such as expert meeting method, pair wise comparison method, Delphi method, etc. No matter applying what kind of method to determine the weight, the weight coefficient of each index must match with the importance degree of each index in index system.

- Form quantization table. Put the designed first, second indicator in the scale, and fill the weight of each indicator under each index item respectively, then it forms a quantitative evaluation (see table 1). Then, you can make use of this table to conduct quantitative evaluation on the information literacy of preschool teachers.

- Make use of fuzzy comprehensive evaluation method to assess the results. Fuzzy comprehensive evaluation method a comprehensive evaluation method which is based on fuzzy mathematics. This approach transfers the qualitative evaluation into quantitative evaluation according the membership theory of fuzzy mathematics. The specific approaches are: Make a real number represents the degree of one object within [0, 1], in which 0 and 1 is the two extreme values of membership. 0 represents the worst and 1 represents the best, other cases locate between 0 and 1. So you can make accurate quantization on the factors which are difficult to quantify.

The Implementation of the Evaluation

Fuzzy comprehensive evaluation is conducted in two steps: First, make a separate evaluation on each factor, and then conduct a comprehensive evaluation on all factors.

The first step: First, make evaluation on single factor.
Take the evaluation on A single factor as an example, set the rating scale set as V, then V = [excellent, good, medium, poor]. The evaluation factors set of A is R, then R = [A1, A2, A3]. From A1, A2, A3 weight proportion in membership table, we know that R = (0.40, 0.40, 0.20). For A factor it adopts M (Λ,V) operator and make evaluation according to the principle of maximum membership degree, then the A single factor evaluation is A = R. V, namely:

\[
A=(0.40,0.40,0.20)
\]

\[
\begin{pmatrix}
0.60 & 0.30 & 0.10 & 0 \\
0.70 & 0.20 & 0.10 & 0 \\
0.80 & 0.20 & 0 & 0
\end{pmatrix}
\]

\[
={(0.40\Lambda0.60)V(0.40\Lambda0.70)V(0.20\Lambda0.80)),((0.40\Lambda0.30)V(0.40\Lambda0.20)V(0.20\Lambda0.20)),
((0.40\Lambda0.10)V(0.40\Lambda0.10)V(0.20\Lambda0)), (0.40\Lambda0)V(0.40\Lambda0)V(0.20\Lambda0))}
\]

\[
=\{(0.40\Lambda0.40V0.20), (0.30\Lambda0.30V0.20), (0.10\Lambda0.10V0), (0V0V0)}
\]

\[
=\{040, 030, 010, 0\}
\]
<table>
<thead>
<tr>
<th>Evaluation factors</th>
<th>First grade index</th>
<th>First grade weight</th>
<th>Second grade index</th>
<th>Second grade weight</th>
<th>Grade</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>A</strong> Information consciousness</td>
<td></td>
<td>0.20</td>
<td>A1 Can actively obtain information.</td>
<td>0.40</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>A2 Can actively learn new information technologies.</td>
<td>0.40</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>A3 Has comprehensive and full understanding of information literacy.</td>
<td>0.20</td>
<td></td>
</tr>
<tr>
<td><strong>B</strong> Information knowledge</td>
<td></td>
<td>0.30</td>
<td>B1 Gets familiar with basic computer operation skills.</td>
<td>0.20</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>B2 Proficiently masters the application of office software.</td>
<td>0.20</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>B3 Proficiently masters the application method of communication tools.</td>
<td>0.20</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>B4 Masters the basic methods of web design and production.</td>
<td>0.15</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>B5 Has the collection and initial processing capacity on multimedia material.</td>
<td>0.15</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>B6 Can be produce multimedia courseware</td>
<td>0.10</td>
<td></td>
</tr>
<tr>
<td><strong>C</strong> Information ability</td>
<td></td>
<td>0.30</td>
<td>C1 Can actively obtain positive information available to solve the problems of education and teaching.</td>
<td>0.30</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>C2 Can make accurate evaluation on information.</td>
<td>0.25</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>C3 Can effectively process, deal with information.</td>
<td>0.20</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>C4 Is willing to share and exchange information with others.</td>
<td>0.25</td>
<td></td>
</tr>
<tr>
<td><strong>D</strong> Information moral</td>
<td></td>
<td>0.20</td>
<td>D1 Does not make use of computer information systems to make, copy, access and disseminate information garbage.</td>
<td>0.60</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>D2 Can actively maintain its own information security.</td>
<td>0.40</td>
<td></td>
</tr>
</tbody>
</table>

Grade description: A means excellent, B means good, C means medium, and D means poor.
Since the sum of a single factor evaluation is 0.40 + 0.30 + 0.10 + 0 = 0.80, the result is not 1. So make normalization we can get (0.40/0.80, 0.30/0.80, 0.10/0.80, 0/0.80) = (0.50, 0.375, 0.125, 0).

By the same way, we can calculate:

The evaluation of single factor B = (0.40, 0.40, 0.20, 0)
The evaluation of single factor C = (0.43, 0.285, 0.285, 0)
The evaluation of single factor D = (0.60, 0.20, 0.10, 0.10)

The second step, make a comprehensive evaluation.

Assuming the grade set of comprehensive evaluation is U, then U = [excellent, good, medium, poor]. Set the factor set of comprehensive evaluation is S, then S = [A, B, C, D]. According to A, B, C, D weight ratio in the membership table, we know that S = (0.20, 0.30, 0.30, 0.20). According to the above calculated results of single factor evaluation, we can get the comprehensive evaluation results:

Comprehensive evaluation = S. U

= (0.20, 0.30, 0.30, 0.20)  
= (0.34, 0.34, 0.32, 0)

The result of comprehensive evaluation shows that 34% of the evaluators think that the information literacy of teachers is "excellent", 34% of the evaluators think that the information literacy of teachers is "good", 32% of the evaluators think that the information literacy of teachers is "medium", no evaluator think that the teacher's information literacy is "poor."

Conclusion

In the evaluation on information literacy of preschool teachers, comprehensive evaluation method which is based on fuzzy mathematics can bitterly solve the problems of how to quantify in evaluation of information literacy. Then this subject can also be analyzed from point of view of data mining. We can apply association rules of data mining to solve the efficiency problem of evaluating data and make in-depth study of information literacy assessment. That is a supplementary to fuzzy mathematics evaluation. So the information literacy assessment of preschool teachers can obtain a more objective and accurate result.

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A RESEARCH ON THE EFFECTIVENESS OF MULTIMEDIA COURSEWARE
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Abstract. The ideological and political theory to bear on the students the task of theoretical education system is the main channel of the ideological and political education. This topic through multimedia courseware in the ideological and political lessons of the classroom to analyze the use of surveys, classroom lessons to improve the effectiveness of ideological and political recommendations.

Keywords: Multimedia, Interactive, Effectiveness.

Introduction
As science and technology and socio-economic development as the core of the computer use of multimedia technology into classroom teaching ever since, he broke the same strain of the rigid state of teaching methods, gives a refreshing feeling. Indeed, the image of multimedia technology, intuitive, vivid, it sound, text, color, shape effect on the eyes, ears, brain and other organs, increase the knowledge capacity, content, fuller, more vivid image, so that information transfer multi-level, multi-faceted, highly scalable and interactive.

According to the 2005 program, ideological and political theory courses from the "1998" program of the seven courses was reduced to four-class colleges, course structure has changed, but the leading role of ideological and political education unchanged. Ideological and political theory is the main classroom teaching positions, I believe that should be on teaching content, teaching methods and means of reforms to improve the ideological and political Classroom Teaching Effectiveness. Today on the ideological and political course of classroom teaching methods to enhance the effectiveness of ideological and political course.

Ideological and Political Department in our hospital after years of all teachers using multimedia teaching practice, and accumulated some experience. We tried through the Ningbo Tianyi Vocational and Technical College, Ideological and Political Survey of Multimedia Teaching, to sum up experiences and lessons learned to solve problems, to maximize the quality of multimedia teaching.

Survey Object
The survey is targeted at our school for ideological and political course of the 2007, 2008, 2009-level multimedia teaching a total of 1685 students. In order for students to express real ideas, to ensure objectivity and accuracy of the survey, this survey taken by secret ballot forms. The survey questionnaires were distributed and 1685 were to recover the 1456 copies, 1456 copies of
valid questionnaires. The survey included students love Ideological and Political Courses, the multimedia courseware evaluation of teaching effectiveness, the process of multimedia teaching problems and the recommendations of students to multimedia courseware in four aspects.

The Survey Data

A. Students in the Degree of Political Course of the Favorite. After entering university, on the ideological and political course of the study, 67.2% of students chose their favorite, while only 12.5% of the students chose not to like, then select the general accounting 20.3 percent, indicating that the students of our colleges and universities concerned with national affairs, concern for social hot spots and focus, with high political and ideological quality.

B. Student Evaluation of Teaching Effectiveness of Multimedia Courseware. The effects of multimedia teaching students choose very satisfied or quite satisfied with 77.4%, 21.8% selected general, only 0.8% of the students choose not satisfied with multimedia teaching; learning effects of multimedia teaching 62.5% of students choose to good effect, the memory profound. Overall, students like on the Ideological and Political Classroom teachers use multimedia courseware for teaching, enthusiasm for the improvement of student learning, the key and difficult to understand and master the content and quality of teaching and sharing the use of resources has a significant role in promoting.

C. Multimedia Teaching Students the Views of the Problems. Teaching students what the current lack of multimedia view, with 50.7% of the students thought the teacher courseware lack of new ideas, forms rigid, only the outline, summary, or diagram-style summary, 30.5% of students believe that information overload, it is difficult to digest in a timely manner, 37.7% of the students also think that teachers operating too fast, not easy to take notes, while 27.1% think that teachers only use of modern multimedia teaching tools to "spoon-feeding" the indoctrination; and multimedia courseware pay more attention to aspects of teaching students to select students pay more attention to accounting for courseware content 9.24%, accounting for animation and film 22.49%, and focus on telling and accounted for 67.07% with the courseware, so most of the students surveyed teachers in the multimedia teaching hospital used in multimedia Courseware is courseware content regardless of the quality, quantity, or the intuitive nature of courseware content, aesthetics positive attitude, but some students think that courseware pictures (including animation) the weight of too little. This is a reminder in the production of courseware for classroom teachers, we should pay attention to increase the graphics, images, animation, video and so on, to make courseware illustrated, lively dynamic forms, and at the same time, strengthen the exchange between teachers and students to enable students to take the initiative to actively participate in and improve classroom teaching.

Analysis of Survey Results

The survey results showed that students of Multimedia Teaching of ideological and political recognition of degrees or high. The survey results also show that the multi-media teaching is to
stimulate student interest in learning an effective method, combined with the actual hospital, multi-media teaching also exist some problems:

A. **The Quality of Multimedia Teaching Facilities.** Multimedia classrooms mainly in large-screen computer projection, poor definition of individual classrooms, classes need to draw the curtains during the day, indoor low light, the students are not used to the new teaching methods, visual fatigue, a high concentration of energy can not be a long time, in the easy to doze off when the lectures. Projector bright enough, pictures are not clear enough, and sometimes sound is not good, the microphone did not catch the sound, some of the equipment processing speed is too slow, and sometimes can not play video courseware. These are the effect of teaching to play.

B. **Teaching the Concept of the Problem.** In the multimedia teaching process, teachers sometimes play like a member of the lack of interaction and communication between teachers and students, teachers, teaching is the inculcation of knowledge, and not enough skilled teachers, computer operations, multimedia courseware for the hypertext features, interactive features, network features lack of awareness, the teaching in the future to be resolved.

C. **Lack of Interaction Between Teachers and Students.** Multimedia courseware is designed by teachers in advance of the teaching process, teaching content or design problems have long been arranged, the process by the teacher controls the classroom, the evolution of multimedia courseware in control. In this environment, students do not think the gap, teachers will only pay attention to the interface between courseware, so that a lack of interaction between teachers and students, bound the divergent thinking of the students, and students in creative thinking and innovative capacity of the teaching purpose is contradictory.

D. **Making the Problem of Multimedia Courseware.** Statistical analysis of the results from the survey, 30.5% of students believe that information overload, it is difficult to digest in time, while 37.7% students thought that teachers operating too fast, not easy to take notes.

**The Appropriate Strategy**

According to the above problems and access to relevant literature, the political course for improving the multimedia teaching my school made the following recommendations for reference:

A. **The Establishment of An Internet Multimedia Classrooms.** The timeliness of the ideological and political features of course, the opening of multimedia classrooms to the Internet, the advantages of using the Internet for classroom teaching, teaching resources, enrich, share and time-sensitive, to maximize its role. At the same time, increase the school's open computer room, students can take full advantage of network resources for learning.
B. The Development of High Quality Courseware. Multimedia courseware is the most direct carrier, is designed for the purpose of teaching computer software, which itself has a teaching, promote learning function. Based on this, first, the school should pay attention to the introduction of courseware, direct introduction of the market has been readily available educational software; the second development of the medical characteristics of the courseware, the medical students of higher vocational college must master the one hand, medical knowledge and skills, and the other On the one hand must have a high ideological and political quality and human quality, the courseware production process to fully reflect this philosophy.

C. The Reform of Teaching Methods. Changes in teaching methods is bound to reform of teaching methods. Multimedia courseware in particular has a strong man - machine - human interaction, this interaction allows students to fully participate in the process of teaching so that students from passive to active learning, students become masters of learning. As a teacher, the need to strengthen multi-media teaching methods under study, and give full play the main role of students and teachers leading role.

D. The School Should Strengthen the Management of Multimedia Hardware. Whether the normal operation of the multimedia, multimedia hardware is the key to strengthen the management of hardware, periodically check the multimedia display screen, projection screen, computer and other facilities, which requires a high level of business management, in the shortest possible time completion of equipment maintenance.

E. To Improve the Use of Multimedia Classroom Environment. Under existing conditions, when most of the multimedia classroom in the use of curtains to cover the needs of light, dark room comparison, in which case the one hand, students could easily lead to visual fatigue, the phenomenon appeared asleep; other hand, the impact of students taking notes easy to cause damage to the eyes.

F. Strengthen Teacher Training in Computer Application. Multimedia teaching is a scientific, technical and art of combining the complex systems engineering, a good production of courseware contains a high level of technology, requires teachers to have a skilled level of computer applications, while the ideological and political non-computer professional teachers, and usually assume the heavy task of teaching, so that in cases of courseware will produce excellent co-operation of school-related departments, regular production of multimedia courseware for teachers training, guidance as well as supervision and provide quality editing tools, build multimedia editing platform, to continuously enhance the use of multimedia courseware for teachers, production level.

Conclusion
In short, questionnaire investigation and analysis, multimedia teaching itself has a huge advantage, ideological and political teaching should give full play to its advantages, growing interest in Ideological and Political Courses students to actively induce the students thinking
about social issues, which greatly improve the ideological and political Classroom teaching effectiveness, to better achieve the purpose of ideological and political education and teaching, so that students become life-long ideological and political lessons of outstanding benefit programs.

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A NEW IMPROVED NEWTON ITERATION METHOD
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Abstract. Firstly, proposed a new Newton iterative formula from the classical Newton iteration method. Then presented the convergence theorem and error equation, and proved that they are at least third-order convergent. If the parameter is chosen properly, the convergence rate will be improved further.

Keywords. iteration, convergence, acceleration.

Introduction
For solving the nonlinear equation, the first classical method is Newton iteration method. The advantage of Newton method is the fast convergence rate, the drawback is the high demand of function derivative, the computation is large, the initial iteration and the exact value are very close. In order to overcome these shortcomings, some Scholars have made improvements to it [1,2,3,4]. The article will building a new Newton iterative formula, in which any non-zero parameters $\lambda$ is selected, the iteration is convergence, proposed convergence theorem and error equation, then proved that it has at least third-order convergence rate, and as long as the proper parameter $\lambda$ selected, you can further convergence speed.

The Improvement of Newton Method
Before the improvement of Newton method, we give some definition. Definition 1: If the iterative formula $x_{k+1} = \varphi(x_k)$ generate Sequences $\{x_k\}$ convergence in $\alpha$, and when $e_k = x_k - \alpha$, Existence of nonnegative real number $p$ and nonzero constant $c$, so

$$|e_{k+1}|/|e_k|^p \rightarrow c (k \rightarrow \infty) \tag{1}$$

Called the iterative formula $x_{k+1} = \varphi(x_k)$ degree of convergence is $p$, and called $e_{n+1} = c e_n^p + o(e_{n+1}^p)$ is error equation.

Traditional Newton iterative scheme is [5]:

$$x_{k+1} = x_k - f(x_k)/f'(x_k) \tag{2}$$

We construct a new iterative formula as follow:
\[ x_{n+1}^* = x_n - \frac{f(x_n)}{\lambda f'(x_n)} \]
\[ x_{n+1} = x_n - \frac{f(x_n)}{\left(1 - \frac{\lambda}{2}\right) f'(x_n) + \frac{\lambda}{2} f''(x_{n+1}^*)} \]

(3)

**Convergence Analysis**

Theorem 1: Suppose \( x^* \) is the single real root of equation \( f(x) = 0 \), \( f(x) \) has fourth-order derivative in \( U(x^*) \), so near by \( x^* \) Newton iterative method (3) is at least at least third-order convergent, its error equation is

\[ e_{n+1} = (c_2^2 + \frac{3c_3}{2\lambda} - c_3)e_n^3 + \left(\frac{3c_2c_3}{2\lambda} + c_2c_3\right)e_n^4 + o(e_n^4) \]

(4)

In which \( e_n = x_n - a \), \( c_k = (1/k)! f^{(k)}(a) / f'(a) \), \( k = 2, 3 \)

Proof: Suppose the single real root is \( a \) of \( f(x) = 0 \), note \( e_n = x_n - a, c_k = (1/k)! f^{(k)}(a) / f'(a) \), \( k = 2, 3 \)

Use Taylor expanded on \( f(x) \) in \( a \), we get

\[ f(x) = f(a) + f'(a)(x-a) + \frac{f''(a)}{2!}(x-a)^2 + \frac{f'''(a)}{3!}(x-a)^3 + o((x-a)^3) \]

Make \( x = x_n \), get:

\[ f(x_n) = f(a) + f'(a)(x_n-a) + \frac{f''(a)}{2!}(x_n-a)^2 + \frac{f'''(a)}{3!}(x_n-a)^3 + o((x_n-a)^3) \]

Derivative (5)

\[ f'(x_n) = f'(a)\left[1 + 2c_2e_n + 3c_3e_n^2 + o(e_n^2)\right] \]

(6)

Use (5) and (6) Substituted into the first Formula of (3), we get:

\[ x_{n+1}^* = x_n - \frac{f(x_n)}{\lambda f'(x_n)} = a + \frac{\lambda - 1}{\lambda} e_n + \frac{c_2}{\lambda} e_n^2 + \frac{1}{\lambda} (2c_3 - 2c_3^2)e_n^3 + o(e_n^3) \]

Use Taylor expanded on \( f'(x) \) in \( a \), we get
\[ f'(x) = f'(a) + f''(a)(x-a) + \frac{f'''(a)}{2!}(x-a)^2 + o((x-a)^2) \]

Make \( x = x_{n+1} \), get

\[ f'(x_{n+1}^*) = f'(a) + f''(a)(x_{n+1}^*-a) + \frac{f'''(a)}{2!}(x_{n+1}^*-a)^2 + o((x_{n+1}^*-a)^2) \]

\[ = f'(a)[1 + \frac{2c_2(\lambda - 1)}{\lambda} e_n + \frac{2c_2^2}{\lambda} + 3c_3(\frac{\lambda - 1}{\lambda})^2 e_n^2 + o(e_n^2)] \]

\[ (1 - \frac{\lambda}{2})f'(x_n) + \frac{\lambda}{2} f'(x_{n+1}^*) = f'(a)[(1 - \frac{\lambda}{2})(1 + 2c_2 e_n + 3c_3 e_n^2 + o(e_n^2)) \]

\[ + \frac{\lambda}{2}(1 + \frac{2c_2(\lambda - 1)}{\lambda} e_n + \frac{2c_2^2}{\lambda} + 3c_3(\frac{\lambda - 1}{\lambda})^2 e_n^2 + o(e_n^2))] \]

\[ = f'(a)[1 + c_2 e_n + (c_2^2 + \frac{3c_3}{2\lambda}) e_n^2 + o(e_n^2)] \]

\[ \therefore \frac{f(x_n)}{(1 - \frac{\lambda}{2})f'(x_n) + \frac{\lambda}{2} f'(x_{n+1}^*)} = \frac{f'(a)[e_n + c_2 e_n^2 + c_3 e_n^3 + o(e_n^3)]}{f'(a)[1 + c_2 e_n + (c_2^2 + \frac{3c_3}{2\lambda}) e_n^2 + o(e_n^2)]} \]

\[ = e_n + (c_3 - c_2^2 - \frac{3c_3}{2\lambda}) e_n^3 + \left(-\frac{3c_2c_3}{2\lambda} - c_3 c_3\right) e_n^4 + o(e_n^4) \]

(7)

The second iterative formula of (3) is:

\[ x_{n+1} = x_n - \frac{f(x_n)}{(1 - \frac{\lambda}{2})f'(x_n) + \frac{\lambda}{2} f'(x_{n+1}^*)} \]

\[ = a + (c_2^2 + \frac{3c_3}{2\lambda} - c_3) e_n + \left(\frac{3c_2c_3}{2\lambda} + c_3 c_3\right) e_n^4 + o(e_n^4) \]

(8)

So by the Definition 1, near by \( x^* \) Newton iterative method (3) is at least third-order convergent. And if parameter \( \lambda \) Selected properly, it will increase Convergence speed further, In particular, when the value of \( \lambda \) make \( c_2^2 + \frac{3c_3}{2\lambda} - c_3 = 0 \), Convergence order will increase at fourth-order.

**Numerical Experiments**

Example 1: Suppose \( f(x) = xe^{x^2} - \sin^2 x + 8, x \in [-5, 5] \), take the iterative initial value as \( x_0 = -3 \), note approximate value as \( l_k \) generated by Newton iterative, make \( \lambda = 1, 2, 8, -2 \) in (3),
the generated approximate value separately as $x_{1k}, x_{2k}, x_{3k}, x_{4k}$, computation result as Tab.1 and Figure 1.

Table 1 Comparison of different iterative methods

<table>
<thead>
<tr>
<th>iterative frequency $l_k$</th>
<th>$x_{1k}$</th>
<th>$x_{2k}$</th>
<th>$x_{3k}$</th>
<th>$x_{4k}$</th>
</tr>
</thead>
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<tr>
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<td>-3</td>
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<td>-3</td>
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<tr>
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<td>-1.79199</td>
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</tr>
<tr>
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<tr>
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<td>-1.79199</td>
<td>-1.57443</td>
</tr>
<tr>
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<td>-1.3011</td>
</tr>
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</tr>
<tr>
<td>7</td>
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<td>-1.30108</td>
<td>-1.30108</td>
</tr>
<tr>
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<td>-1.30484</td>
<td>-1.30108</td>
<td>-1.30108</td>
</tr>
</tbody>
</table>
Fig. 1 Comparison of different iterative methods.

From Table1 and Figure 1, we can easily observed, if parameter $\lambda$ Selected properly, the Convergence speed of the Newton iterative method proposed by this article is much faster than classical Newton method and improved Newton method by some literature. However, we did not calculate which $\lambda$ value, the convergence speed is the fastest, it is needed further research and discussion.

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RECTIFICATION METHODS ON DISTORTION OF DIGITAL AIRBORNE IMAGES
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Abstract. This paper mainly describes the rectification method of digital airborne image distortion. First, according to the digital sensor model, the mathematical model of airborne image is built up for processing the geometrical data of image, which is also the key for digital rectification. Then, the principle of digital rectification is explained through discussing the direct method and the indirect method in detail. Because the coordinates of image points are not usually set in the center of pixel, the bilinear interpolation is used for gray resampling in the indirect method, which can obtain the rectified images with high accuracy. Finally, according to the theory and the method provided in the paper, program of digital airborne image rectification is coded by Visual C++ language. On the analysis of experimental results between direct method and indirect method, the rectification method for distortion of digital airborne image is proved to be practical and correct in the real applications.

Keywords: Digital Airborne Images, Rectification of Distortion, Mathematical Model, Digital Photogrammetry.

Introduction
With the development and the advance of digital photogrammetry, the meaning and the contents of airborne photogrammetry have been constantly added and improved. Its application fields and ranges are continuously expanding and enlarging along. Especially in the past decade, great and urgent demand for reliably obtaining and rapidly updating the spatial data have effectively promoted the technologies and equipments of airborne photography with a high-speed development and perfection. Therefore, the key of airborne photography lies in changing the way to acquire the target information, which is from the photosensitive imaging mode of past to the recording digital recording mode of present. In another words, the main equipment is transferred the optical aerial camera into the digital aerial camera.

The digital airborne images with high resolution are applied in the fields of linear mapping, orthophoto making, 3D reconstruction, navigation monitoring, hydrological analysis and so on. In addition, the digital images are also applied in the integration of spatial information and real-time GIS data, such as natural disaster monitor and forecast, soil range change and city reconstruction.

For a long time, the original information data of photogrammetry are carried from photographic film or plate. The digital data of photogrammetry are also formed by digital scanning. However,
the digital airborne images can be directly shot by airborne digital camera, which greatly improves the efficiency of photogrammetry with high spatial and radiation resolution. The airborne digital camera is imaged by push broom with line central projection, and its CCD detector is affected seriously by flight motion and other factors in the period of sampling. Thence, the original images are obtained with big deformation and distortion which would affect not only the image visualization but also stereoscopic observation and match in the next step. Therefore, the rectification of digital airborne image is one of the most important issues for further applications.

Model of Digital Airborne Image

Model Build. In order to analyze digital images, it is usually set down that X direction is the direction of sensor scanning and Y direction is the flight direction. In the plane coordinate system, the exterior orientation parameters are supported as $X_s$, $Y_s$, $Z_s$, $\phi_i$, $\omega_i$, $\kappa_i$, and the imaging equation is shown in the equation 1.

$$
\begin{align*}
x - x_0 &= -f \frac{a_1(X - X_s) + b_1(Y - Y_s) + c_1(Z - Z_s)}{a_3(X - X_s) + b_3(Y - Y_s) + c_3(Z - Z_s)} \\
y - y_0 &= -f \frac{a_2(X - X_s) + b_2(Y - Y_s) + c_2(Z - Z_s)}{a_3(X - X_s) + b_3(Y - Y_s) + c_3(Z - Z_s)}
\end{align*}
$$

(1)

Among the equation 1, $x$ and $y$ are the coordinates of image point, and $x_0$, $y_0$, $f$ are the interior orientation parameters of image, and $X_s$, $Y_s$, $Z_s$ are the spatial coordinates of camera, and $X$, $Y$, $Z$ are the spatial coordinates of object, and $a_i$, $b_i$, $c_i$ ($i = 1, 2, 3$) are 9 parameters of 3 exterior orientation parameters of image forming a rotation matrix shown in the equation 2 below.

$$
R = R(\phi)R(\omega)R(\kappa) = 
\begin{bmatrix}
a_1 & a_2 & a_3 \\
b_1 & b_2 & b_3 \\
c_1 & c_2 & c_3
\end{bmatrix}
$$

(2)

$$
\begin{align*}
a_1 &= \cos \phi \cos \kappa - \sin \phi \sin \omega \sin \kappa \\
a_2 &= -\cos \phi \sin \kappa - \sin \phi \sin \omega \cos \kappa \\
a_3 &= -\sin \phi \cos \omega \\
b_1 &= \cos \omega \cos \kappa \\
b_2 &= \cos \omega \sin \kappa \\
b_3 &= -\sin \omega \\
c_1 &= \sin \phi \cos \kappa + \cos \phi \sin \omega \sin \kappa \\
c_2 &= -\sin \phi \sin \kappa + \cos \kappa \sin \omega \cos \kappa \\
c_3 &= \cos \phi \cos \omega
\end{align*}
$$

(3)
Because of this equation in the derivation process, image moments like the corresponding feature point should be located in a straight line through the sensor's center, it is also known as the collinear equation (referred to as the collinear equation), it is built on the basis of simulation and calculation of position and attitude sensor imaging process, reaction like the plane like the correspondence between points and the ground point coordinates, the image plane coordinate system and the object space coordinates together, for aerial image geometric deformation analysis and digital correction provides theoretical basis.

**Coordinate Transformation.** The digital airborne image reflects the relation between image point of image plane and spatial point of earth surface. The geometric rectification of image is just to establish the correct relation of both above. In this process, there are involved several coordinate system transformations which are original image plane, image plane, spatial plane and rectified image plane, as shown in the Figure 1.

![Fig. 1 Transformation of coordinate systems.](image)

The original image coordinates for scanning coordinate system, it should use the original image scanning coordinates and plane coordinates as the mapping relationship between the two, respectively, and then obtain the corresponding image plane coordinates. In fact, from the original image to the image plane coordinate conversion, is the process of linear interpolation. From the image plane to the coordinate transformation in object space collinearity condition equation is mainly based on. Coordinate transformation from object space to correct the image by scaling, rotating, peace in description. Coordinate to achieve the correct image to object space conversion, must have the angle of rotation, scale and offset parameters, these parameters in the correct time, or given directly, or through the calculation.

**Rectification Distortion of Digital Airborne Image**

According to the error sources and their influences, the geometric rectification of image is divided into the optical rectification and the digital rectification. The basic form of optical rectification is to reverse photographic process by means of correcting instrument, which is a traditional method of rectifying image in photogrammetry. With the development of computer technology, the geometric rectification of airborne image has transferred from the optical or mechanical method to digital method. The applications of digital image processing technology not only strengthen image and adjust contrast but also form digital differential rectification.

**Direct Rectification Method.** The Principle of direct rectification method is shown in the Figure 2. The method starts from the point of view of each pixel of the original image, and calculates
the correct position of pixel in the output image according to the order of each primitive point with positive solutions of equation 4. After the coordinates of image points are calculated correctly, the original images can be rectified completely.

\[
\begin{align*}
X &= Z \cdot \frac{a_1 x + a_2 y - a_3 f}{c_1 x + c_2 y - c_3 f} \\
Y &= Z \cdot \frac{b_1 x + b_2 y - b_3 f}{c_1 x + c_2 y - c_3 f}
\end{align*}
\] (4)

Among the equation 4, \(x\) and \(y\) are the image coordinates of original images, and \(X\) and \(Y\) are the corresponding image coordinates of rectified images.

**Indirect Rectification Method.** The indirect rectification method is different from the direct rectification method above. It starts from the blank rectified image and calculate the original image pixel coordinates in the reversed position. Its principle is shown in the figure 3.
Experiments and Results

According to the mathematical model mentioned above, this experiment adopted the correct program in Visual C++ language. The program realizes to rapidly display the digital airborne images with massive high resolution and to determine the correct parameters and fixed heights. The geometric rectification of original images by direct method and indirect method are compared and analyzed from the experimental data and results, which is shown in the Figure 4.

Original image with geometric distortion is shown in the Figure 5 (a). According to the digital rectification method mentioned above, the rectified images are shown in the Figure 5 (b) and (c).
Conclusion

The rectification of image distortion is an important premise for photogrammetric processing. Because the original images with geometric distortion cannot be directly measured and located with aerial triangulation and orientation. Only the rectified images can be used and applied for the next operations. Therefore, the digital rectification of original images is necessary and
required entirely. The experimental results show that the rectification method is effective and practical for image airborne images with high precision in the paper.

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References


A DIGITAL COLLABORATIVE DESIGN MODEL BASED ON LEARNER CHARACTERISTICS IN ART DESIGN

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Abstract. Personalized design system can provide better service for learners in art design. The digital collaborative learning model is proposed with adaptive design contents, design strategies and grouping according to learner characteristics. The Definition and formal description of design model in digital design system is given, collaborative design model is designed and a collaborative strategy is generated.

Keywords: Art design, Digital design system, Collaborative design mode, Learner characteristic.

Introduction

With the rapid development of information technology, the personalized learning system is already a hotspot in digital design and intelligent education. The aim of digital design system is how to provide better personalized service for learners in art design. If learners with the same mode, it can not satisfy the learner with his personalized need. But if the learner model makes initial estimations of the new learner’s knowledge level and his personalized characteristics before his study, the problem of personalized need will be solved. So the research of learner model is the primary problem in digital design.

At present, several generic norms such as the AGR (AICC Guidelines and Recommendations), SCORM (The Sharable Content Object Reference Model) and IMS (Instructional Management Systems) are committed to solve the two major issues of resources sharing and interoperability, and have achieved satisfactory results. [1] In terms of learner model, typically included vector model, overlay model and fault model. [2] In vector model, Learner’s knowledge was modeled in a vector consisted of concepts or topics or subjects. Each element in the vector shows the degree which learner gains knowledge about those concepts, topics or subjects. [3] In overlay model, Learner’s knowledge is the subset of expertise’s knowledge. Each element in overlay model is the number which presents learner’s knowledge level. [4] Fault model can contain learner’s errors and the reasons of the errors. [5] The drawback of vector model overlay model and fault model is that it can not describe the lack of learner’s knowledge. Then, it is poor that the research results of collaborative design model according to the learner characteristics. So the digital collaborative model is proposed for the learner’s personalized design in art design.
The paper is organized as follow: In section 2 Definition of Learner Model included Formal Description and Components of Learner model is introduced. In section 3 details about the digital collaborative design model. Finally, the conclusion is proposed.

**Definition of Learner Model**

**A. Formal description of Learner model.** A learner model is defined as “a representation of the learning system’s beliefs about the learner”. Accord to the analysis results of learner model in the digital design system, It include Learner Profile, design Level, Cognitive Ability, Learner Style and Learner Motivation. They respectively defined as followed:

Learner model := { Learner Profile, Knowledge Level, Cognitive Ability, Learning Style, Learner Motivation}

Learner Profile := {StuNo, Name, Gender, Age, Nation, Major, Email}
Design Level := { 1 low level, 2 medium level, 3 high level}
Cognitive Ability := { Know, Understand, Apply, Analyze, Synthesize, Evaluate}
Learner Styles := { active vs. reflective, sensing vs. intuitive, visual vs. verbal, sequential vs. global.}
Learner Motivation := { Attention, Relevance, Confidence, Satisfaction}

Attention := { 0 Additional support/Low attention, 1 a bit additional support/in attention , 2 No additional support/Higher attention}
Relevance := { 0 Additional support/Low relevance, 1 a bit additional support/in relevance , 2 No additional support/Higher relevance}
Confidence := { 0 Additional support/Low confidence, 1 a bit additional support/in confidence , 2 No additional support/Higher confidence }
Satisfaction := { 0 Additional support/Low satisfaction, 1 a bit additional support/in satisfaction , 2 No additional support/Higher satisfaction }

**B. Components of Learner model**

1) **Learner Profile.** Learner profile provides learners the basic characteristics in order to distinguish the different learners and their professional background. Learner Profile information is static information, in the whole learning process will not change.

2) **Design Level.** Before learners begin to learn, the system offers a pre-test paper according to the learning contents which can evaluate the learner’s present knowledge level. The result of pre-test is the grade of learner’s knowledge level. Learner’s knowledge level is dynamic, so the fuzzy set of the learner in different knowledge level will be divided into three levels: low, medium, and high, respectively representd with 1, 2 and 3. The relevant degree of every level is expressed as $\mu_i(\bar{A})$. A is the fuzzy set is expressed as:

$$\bar{A} = \sum_{i=1}^{3} \frac{\mu_i(\bar{A})}{i}$$

in:

$$0 \leq \mu_i(\bar{A}) \leq 1, \sum_{i=1}^{3} \mu_i(\bar{A}) \leq 1, i=1,2,3$$

The princilpe of the determining set is the maximum relevant degree of learner knowledge level. For example, the three relevant degrees of a learner’s knowledge level are 0.1, 0.7, and 0.2), then the fuzzy set of knowledge level $\bar{A} = 0.1 / 1 +0.7 / 2 +0.2 / 3$. According to A’s expression, the level 2 is highest in relevant degree, so the learner’s knowledge level is medium. If two learners are at the same level in relevant degree, they are similar in the knowledge level.

3) **Cognitive ability.** In accordance with Bloom’s “Taxonomy of educational objective” theory in this system. Every learner’s cognitive ability corresponds to a vector which has six
dimensions that is {know, understand, apply, analyze, synthesize and evaluate}. The fuzzy comprehensive evaluation theory is used to obtain the learner cognitive abilities. The sub-cognitive level of each dimension: Low [0,60], medium [60,80], high [80,100], and each dimension of cognitive ability given different weights, and the sum of all weights is 1, with the formula is:

\[ r_i = \sum_{k=1}^{6} (r_{ik} \cdot w_k) \]  

(2)

\( r_{ik} \) is the cognitive ability of the learner \( l_i \) in the k dimension, \( w_k \) is the weight of, and \( k \) is the dimension, \( k = 1, 2 \ldots 6 \).

The learners in the same group are similar in cognitive ability, is better to help and promote each other. According to the similarity of cognitive ability, the learner \( l_i, l_j \) can or not be assigned in the same group, the following should is considered:

(1) \( l_i \) and \( l_j \) belong to the same level in cognitive ability. Learners should be allocated in the same group.

(2) They belong to different levels of cognitive ability. (a) the learner \( l_i \) is a high (low) levels, \( l_j \) is low (high) level, learners \( l_i, l_j \) should be allocated in different groups. (b) the learner \( l_i \) is a high (low) level, \( l_j \) is medium level, or learners \( l_i \) is medium level, \( l_j \) is high (low) level, computing \( l_i \) and \( l_j \) the similarity of cognitive ability \( s(l_i, l_j) \), the formula as follows:

\[ s(l_i, l_j) = \sum_{k=1}^{6} |r_{ik} \cdot w_k - r_{jk} \cdot w_k| \]  

(3)

\[ r(l_i) = \sum_{k=1}^{6} (r_{ik} \cdot w_k) \quad r(l_j) = \sum_{k=1}^{6} (r_{jk} \cdot w_k) \]

in:

If \( s(l_i, l_j) < \beta \) \((0 \leq \beta \leq 5)\), then the learners \( l_i, l_j \) is similar in cognitive ability, so, the learners \( l_i, l_j \) is located in the same group; Otherwise, they is assigned to different groups.

4) Learner style. Typically, the scale is used to test learner style. The learner style is divided into field independence (marked as: 1) and field dependence (marked as: 0). To not or rarely affected by environmental factors is known as field independence, to those affected by major environmental factors is known field dependence. Field independence is often good at analysis and organization, easy to accomplish those who need to identify the key elements of the task and re-organization of materials; Field dependent tends to focus on one aspect of the scenario, a model can not be broken down into many parts. Collaborative design emphasizes the learner to communicate with others, collaboration skills. Measuring learner styles in the form of tables to the web form is given back by the system after the submission of the proposed algorithms to determine what learners are learner styles.

5) Learner motivation. Keller (1979, 1983) has developed a four-factor theory to explain motivation. The four-factor is attention (A), relevance (R), confidence (C), and satisfaction (S). The model also contains strategies that can help an instructor stimulate or maintain each motivational element. A learner’s attention has to be aroused and sustained. This category also includes things that relate to curiosity and sensation seeking. After the learner’s attention is gained, he may wonder how the given material relates to their interests and goals. If the content
is perceived to be helpful in accomplishing one’s objectives, then they are more likely to be motivated. Learners will know that they will probably be successful before completing a given task. They will feel somewhat confident. If the outcome of a learner’s effort is consistent with their expectations and they feel relatively good about those outcomes, they will remain motivated.

**Collaborative Design Model**

According to the learner’s design needs, design level learner style, the system actively adjusts the design content, learner style, design strategy, learning paths and design support, making the difficulty of design resources suited to the learner’s level, the presented design content suited to the learner needs, and the provided guidance suited to the learner’s design strategies. The whole design process is learner-centered, personalized to meet the learner’s design needs, in which the learner’s dominant position if fully realized. The self-construction of knowledge, as shown in Figure 1.

Learners actively interact with collaborative design system, constantly obtain, analyze and feedback information, so as to construct their knowledge. By the design system recording of the design process, learners can timely control and adjust their own design process to achieve optimized objectives suitable to them. The design system is intelligent. Intelligence is the basic guarantee for the system to realize self-adaptation. It enables the system make comprehensive and scenically diagnose of the learner’s actual level and psychological conditions, present suited design content and design support according to the presetting and design process tracking.

![Fig. 1 Digital collaborative design model.](image)

**Conclusions and Future Work**

The collaborative design model is proposed with adaptive design contents, design strategies according to personality characteristics. According to the formal description of learner model in digital design system, then the collaborative design model is proposed.
In the future, the research in learner model should include other factors, such as social preferences, learning EQ, and so on. The research should apply intelligence algorithm to support more complex collaborative design model in art design. The algorithm of mining learner characteristics in design model would continue to perfect.

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