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Adaptive Control of the Radial Servo System of a Compact Disc Player

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Abstract

An important research area for modern CD-based consumer electronic devices is to find a way to enhance the behaviour of the controlled system by increasing the bandwidth. If this is possible then the disturbance rejection improves, which is an important issue for CD-players with portable use. With regard to the field of data storage systems one can then also increase the rotational frequency of the disc, to increase the data readout speed, as well as decrease the track size, to increase the data density on the disc. However, pushing up the bandwidth of this system introduces the problem of high frequency resonances, caused by certain bending and torsional modes, to have more influence on the closed loop behaviour. Due to production tolerances, these high frequency dynamics vary from player to player. This high frequency dynamical variation, as well as the fact that the system has a varying process gain during operation explains the need for flexibility in the controller.

In this respect, this report considers the design of a flexible controller using adaptive techniques for the radial servo system of the rotating arm Compact Disc mechanism CDM-9. Applying these techniques to this problem, makes it possible to compensate for dynamical variations on-line.

The most general adaptive scheme of the self tuning regulator (STR) is used to compensate for the effects of one resonance peak as well as the varying process gain. The adaptive controller designed in this way is used in combination with a fixed controller designed with the technique of pole-placement design. Simulations show that the controller designed as such can cancel the effect of one resonance peak sufficiently. It is also able to control the varying process gain over the entire variation range.

The most severe disadvantage of the designed adaptive scheme was the high complexity of the estimation stage of the STR. The high complexity and need for many calculations will give problems with regard to the short computation time restricted by the high sampling frequency.

Key words
adaptive control, self tuning regulator, pole-placement design, recursive parameter estimation, compact disc player

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Contents

1. Introduction

1.1 About this report

This report describes the design of a flexible controller for a rotating arm Compact Disc mechanism using adaptive techniques. Besides a short overview of the various techniques available in adaptive control, it comprises a design of the controller together with simulation results.

The research is carried out at the Department of Automatic Control of the Lund Institute of Technology, Sweden, as a preliminary report for the MSc. thesis project for Delft University of Technology, Department of Mechanical Engineering and Marine Technology.

1.2 Motivation

Since the introduction of the CD-player in 1981 its popularity has increased rapidly. Within a few years it had outmarketed the traditional record player and it has now found its use in other applications. Apart from the well known "in-home" CD-player there now also exists the portable kind, such as the portable-CD, the car-CD and the player which is found in so-called sound-machines. It has also found its use in the field of computers, for instance the CD-rom and the optical digital drive (ODD).

These new applications ask higher performances from the system than the "in-home" player. The portables have to be less sensitive to external disturbances, such as shocks or shaking. For the computer applications one would like to see a higher rotational frequency of the disc to increase the data readout speed or smaller tracks on the disc to increase the data density.

To achieve these new specifications pushing up the bandwidth of the system is a good solution. However, when raising the bandwidth one runs into the problem of high frequency resonance peaks. These are caused by bending and torsional modes present in the system.

The presence of these high frequency dynamics does not have to be a problem if the shape and the frequency of the resonance peaks are known. However, because of production tolerances these dynamics vary from player to player. In order to push up the bandwidth it is thus necessary to design a controller that can handle the problem of these varying dynamics. There are several solutions that can be found to handle this problem. However, when designing a controller for this system one has to keep in mind that the CD-player is a low cost consumer electronic. This means that the cost of such a system must be kept as low as possible in order to stay competitive. This poses restrictions on the complexity of the designed system.

One solution is to design a fixed controller that is robust in the face of the varying dynamics. The H_∞-theory is then used. This technique provides the possibility to trade-off robustness and performance requirements on a structural basis. The application of the robust control theory to this problem can been found in [5]. The order of the resulting controller design using this technique was too high to be actually implemented in the CD-player.

Another possibility is to design a flexible controller that can be tuned or adapted to the varying dynamics. With the tunable option the tuning is done once and then the tuned fixed controller is placed into the system. With the adaptive variant the adaptive system is built into the player and can tune the controller to the varying dynamics on-line. This method is particularly interesting if the dynamics of the system change during operation. The advantage of such a control system over the robust controller is that the conservatism in the design will be smaller. This is because the amount of unknown dynamics is less.

It is the intention of this report to investigate possible improvements of the track-following behaviour of a Compact Disc player, using adaptive control techniques.

1.3 Organization of chapters

Adaptive control is a field of control design which is extensive and has a lot of different theories and applications. Chapter 2 will therefore give a short overview and a presentation of adaptive control. This chapter will also give a more detailed description of the adaptive method which is used for this problem, the self tuning regulator.

The problem which has to be solved will be described in detail in Chapter 3. Different aspects of the problem will be treated such as specifications, disturbances and complications that are inherent to the CD-problem.

Chapter 4 will be devoted to control design. The design will be split into two parts. First the design of the fixed controller is explained. Then the design of the flexible part of the controller based upon adaptive techniques will be discussed.

Chapter 5 will show some simulation results. Finally the last chapter will give an evaluation of the used method and the final design of the controller.

2. Adaptive control

The book Adaptive Control [8] does not give a final definition of the subject adaptive control. It only provides the following description :

- Adaptive Control is a special type of nonlinear feedback control in which the states of the process can be separated into two categories, which change at different rates. The slowly changing states are viewed as parameters. This introduces the idea of two time scales: a fast time scale for the ordinary feedback and a slower one for updating the regulator parameters. This implies that linear constant parameter regulators are not adaptive. In an adaptive controller we also assume that there is some kind of feedback from the performance of the closed-loop system.

2.1 Overview of various techniques

There exist different methods for designing an adaptive system for a plant. In the following paragraphs a short description will be given of the most important methods together with their strong and weak points.

Gain scheduling

In some systems there are auxiliary variables that relate well to the characteristics of the process dynamics. If these variables can be measured they can be used to change the regulator dynamics. This approach is called *gain scheduling.* In figure 2.1 a diagram is shown of the basic idea of gain scheduling (figures 2.1, 2.2 and 2.3 are taken from [8], courtesy of K.J.Astrom and B.Wittenmark).

Gain scheduling is an open-loop compensation and can be viewed as a system with feedback control in which the feedback gains are adjusted by feedforward compensation. There is no feedback from the performance of the closed-loop system. With regard to nomenclature, it is therefore controversial whether this method should be considered as an adaptive system or not, because the parameters are changed in open loop.

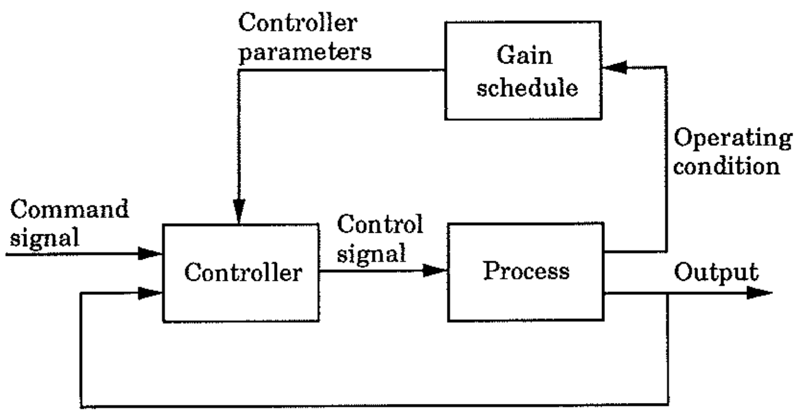

Figure 2.1 Blockdiagram of a system with gain scheduling

Auto tuning

Simple PID controllers are adequate in many applications. Such controllers are traditionally tuned using simple experiments and simple empirical rules. Many adaptive techniques can be applied to tune PID controllers.

Tuning is usually based on an experimental phase in which test signals such as steps or pulses are injected. The regulator parameters can be determined from the experiments using standard rules for tuning PID controllers.

An advantage of auto-tuners is that the tuning experiment is initiated and can be supervised by an operator.

Model-reference Adaptive Systems (MRAS)

The *model-reference adaptive system* was originally proposed to solve a problem in which the specifications are given in terms of a reference model that tells how the process output ideally should respond to the command signal.

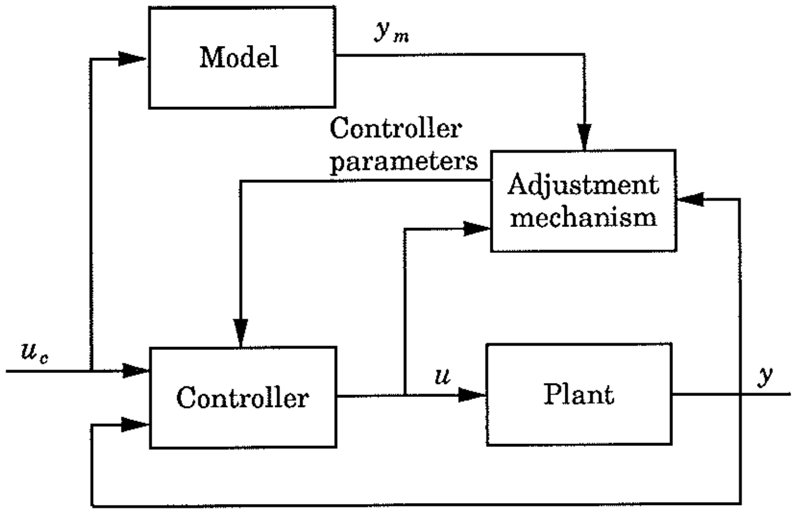

Figure 2.2 Blockdiagram of a model-reference adaptive system (MRAS)

The regulator can be thought of as consisting of two loops: an inner loop, which is an ordinary feedback loop composed of the process and the regulator. The parameters of the regulator are adjusted by the outer loop in such a way that the error e between the process output y and the model output y_m becomes small (figure 2.2). The key problem with this method is to determine the adjustment rule in such a way that the system remains stable. The MRAS attempts to adjust the parameters in such a way that the correlation between the error e and the sensitivity derivatives of the error with respect to the adjustable parameters θ becomes zero.

An example of an adjustment rule is the *MIT-rule*:

$$\frac{d\theta}{dt} = -\gamma e \frac{\delta e}{\delta \theta}$$

In this equation e denotes the model error. The components of the vector $\frac{\delta e}{\delta \theta}$ are the sensitivity derivatives. The parameter γ determines the adaptation rate.

If the MIT-rule is applied to very simple problems, such as adaptation of a feedforward gain then no approximations of sensitiviy derivatives are needed. However, when the problem gets more complicated approximations will have to be made.

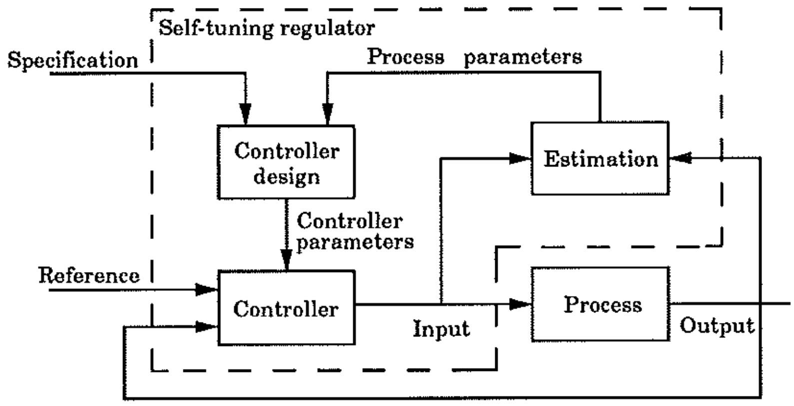

Figure 2.3 Blockdiagram of a self-tuning regulator (STR)

Self-Tuning Regulators (STRs)

The schemes discussed so far are called direct methods, because the adjustment rules tell directly how the regulator parameters should be updated. A different scheme is obtained if the process parameters are updated and the regulator parameters are obtained from the solution of a design problem. This method can be considered as having the same two loops as the MRAS. The inner loop consists of the process and an ordinary linear feedback regulator. The parameters of the regulator are adjusted by the outer loop, which is composed of a recursive parameter estimator and a design calculation. This system can be seen as an automation of process modeling and design, in which the process model and the control design are updated at each sampling period. A controller of this construction is called a *self-tuning regulator (STR)* (figure 2.3).

The STR scheme is very flexible with respect to the choice of the underlying design and estimation methods. It can also handle more complex problems without running into very large complications. The regulator parameters are updated indirectly via the design calculations. It is sometimes possible to reparametrize the process so that the model can be expressed in terms of the regulator parameters. This step eliminates the design calculations which simplifies the algorithm.

Relations between MRAS and STR

It appears from the descriptions of MRAS and STR systems that they are closely related. Both systems have two feedback loops. The inner loop is an ordinary feedback loop. The regulator has adjustable parameters, which are set by the outer loop. The adjustments are based on feedback from the process inputs and outputs. However, the methods for design of the inner loop and the techniques used to adjust the parameters in the outer loop are different. The regulator parameters are updated directly in the MRAS and are updated indirectly via parameter estimation and design calculations. This difference is not fundamental, because the STR may be modified so that the regulator parameters are updated directly.

2.2 Used theory

The method used to solve this problem is the adaptive scheme of the self tuning regulator (STR). It is chosen because of the fact that this method can handle more complex problems than the other methods. It is also very flexible with respect to the choice of the underlying design and estimation method. This section will give a short overview of the used theory.

Self tuning regulators

Development of a control system involves many tasks such as modeling, design of a control law, implementation etc. The self-tuning regulator attempts to automate several of these tasks. Parameters of the model are estimated on-line. The estimation is done using a recursive estimation method. The controller is then redesigned using the estimated parameters.

The different tasks of this method, estimation and controller design, can be performed in many different ways. For instance, the estimation can be peformed either continuously or in batches. The estimation method itself can also be chosen: stochastic approximation, least squares, extended and generalized least squares etc. There is also a wide variety of methods which can be used for control design such as PID, minimum variance control, linear quadratic control or pole-placement design.

The estimation method applied in this report is the recursive least squares estimation method (RLS) [8],[7]. This method updates the parameters every sampling period.

The controller is split into two parts. One part is fixed, the other is one is an STR. The control design method for the fixed part of the controller will be the pole-placement design method described in [8]. The control design for the flexible part will be one in which the goal will be to cancel the identified dynamics completely. Details about this design method can be found in section 4.3.

The method of pole-placement design for the fixed part of the controller is chosen because this method facilitates the redesign of the controller if the desired closed loop bandwidth is changed. It turns out that the final controller designed with this method was not at all different from controllers designed with other methods carried out recently. In the next section a short description of the pole-placement design method will follow.

Pole-placement design

The idea of the pole-placement design method is to determine a controller which gives desired closed loop poles. Further more it is required that the response of the closed loop system to command signals satisfies a model following condition.

The process is described by the single-input-single-output system

$$A(q)y(t) = B(q)(u(t) + v(t))$$

where y is the output, u is the input of the process and v is the noise which enters at the process input. The A and B-polynomials are polynomials in the forward shift operator. The degrees are $degA = n$ and $degB = n - d_o$. The term d_o is called the pole excess. It is assumed that A and B are relatively prime and A is monic.

A general linear regulator can be described as:

$$Ru(t) = Tu_c(t) - Sy(t)$$

This represents a controller with two degrees of freedom; the feedback operator $-\frac{S}{R}$ and the feedforward operator $\frac{T}{R}$.

For the closed loop system then follows

$$y(t) = \frac{BT}{AR+BS}u_c(t) + \frac{BR}{AR+BS}v(t)$$

The closed loop characteristic polynomial is thus:

$$AR + BS = A_c \tag{2.1}$$

The key idea is now to specify the desired closed loop characteristic polynomial A_c. The polynomials R and S can then be solved from 2.1. Equation 2.1 is called the Diophantine equation. It always has solutions if A and B have no common factors. The solutions can be obtained by introducing polynomials with unknown coefficients and then solving the linear equations obtained as such.

With the diophantine equation only the polynomials R and S are determined. Other conditions must be introduced to determine the polynomial T. This can be done by requiring that the response from u_c to the output be described by the dynamics:

$$A_m y_m(t) = B_m u_c(t)$$

This is the previously mentioned model following condition. The following condition must then hold:

$$\frac{BT}{AR+BS} = \frac{BT}{A_c} = \frac{B_m}{A_m} \tag{2.2}$$

Equation 2.2 implies that there are cancellations of factors of BT and A_c. To avoid badly conditioned cancellations we factorize B as:

$$B = B^+B^-$$

where B^+ is monic and contains the part of B that can be cancelled. The polynomial B^- contains the unstable and badly damped modes. The diophantine equation then reduces to:

$$AR' + B^-S = A_oA_m$$

with $R' = \frac{R}{B^+}$ and A_o is the observer polynomial. The polynomial A_o is called the observer polynomial because of the analogy that can be found between this polynomial and the observer used in general feedback problems.

Some conditions must be imposed on the controller polynomials so that it is causal in the discrete time case and proper in the continuous time case:

$$degS \leq degR$$

$$degT \leq degR$$

The diophantine equation has many solutions. However, the solution with the lowest degree is desired. This is called the minimum degree solution. The controller with the lowest degree can be found using

$$degR = degA_c - degA$$

The condition $degS \leq degR$ implies

$$degA_c \geq 2degA - 1$$

The condition $degT \leq degR$ implies

$$degA_m - degB_m \geq degA - degB = d_o$$

With these inequalities the degree of the various polynomials of our problem can be found. The diophantine equation can then be solved, obtaining as such the parameters of the controller.

3. Description of the problem

Now a detailed description of the problem will be given. First the CD-mechanism will be described. Subsequently a model of the system will be made. Together with a block diagram of the system this should give a good view of the different subsystems and signals. Other aspects such as specifications and disturbances will be investigated. Finally two problems that are inherent to this CD-problem, the varying process gain and the varying high frequency dynamics, will be discussed.

3.1 Description of the CD-mechanism

The mechanism that is considered in this report is the CDM-9, a rotating arm mechanism. Schematically the CD-mechanism is given in figure 3.1 (courtesy of R. de Callafon).

A compact disc is a reflective disc containing digital information on a spiral

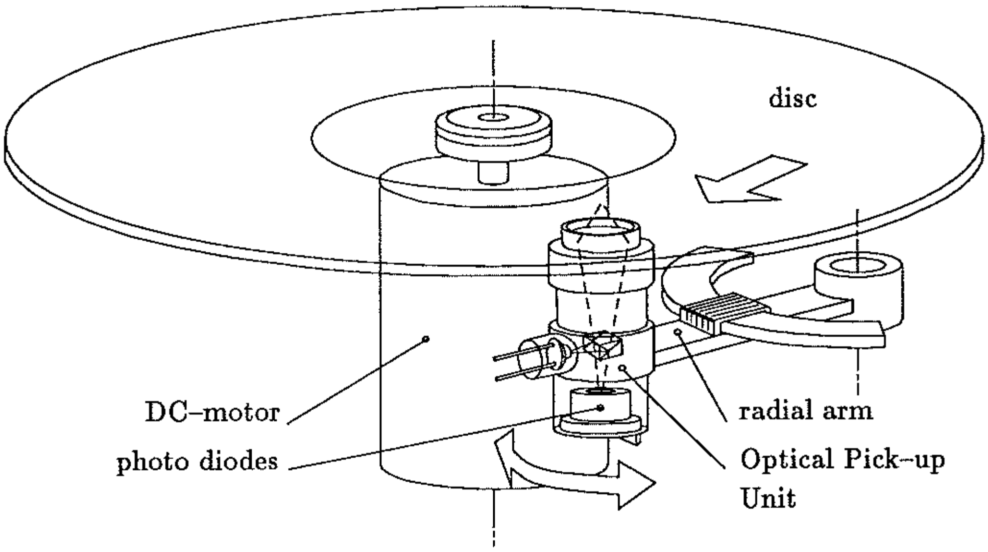

Figure 3.1 Schematic view of CD-mechanism

shaped track [2]. The information is stored in the tracks in the form of pits of various lengths. One can distinguish the pits because of their discrete distribution in length. The bit-length is equal to $0.3\mu m$. The width of the track is $1.6\mu m$.

The information on the disc is retrieved by using a focused laser. The laser is projected after passing through several lenses and is reflected by the disc. The normal disc level and the pits give different intensities of light reflection. The reflected laser light passes back through the lenses and is reflected to a light sensitive diode system by a semitransparent mirror. There is therefore no contact between the pickup device and the information carrier.

Apart from the audio information the tracks also carry other information. There is information about timing and error decoding and the system reads sync-bits for speed control. The latter are needed due to the fact that the readout speed must be kept at a constant value. That means that the rotation speed must vary as the pickup moves from the inside to the outside of the disc. This rotation speed varies between 4 and $8Hz$. This type of rotation is called Constant Linear Velocity (CLV). The control system that controls the rotation speed works in the following way. The readout speed is determined by comparison of the rate of incoming data from the disc with a reference frequency generated by a quartz pulse. The data is read into a FIFO buffer. The difference between the pointer position of the incoming data and the pointer position of the outgoing data is used as a control error signal for the turn-table control system. Therefore the actual rotation speed is not known. Only the error between the actual and required speed is available [5].

The signal the diode system produces is the error between the track and the spot position and a similar signal for the focusing action. This means that the actual spot position and the reference signal are also unknown. The error signals are fed into the controller which in its turn generates the control signal for the tracking and the focussing servo.

For access procedures, i.e. jumping from one track to another, a different control strategy is applied. This strategy presupposes high quality track following, as discussed here, but its details are beyond the scope of this paper.

The laser spot position can be controlled by means of two actuators; the radial and the focus actuator. The radial actuator, which controls the track-following, consists of a moving coil which moves around a bended permanent magnet. This system can be considered to behave as a double integrator. The input is the current i_{rad} to the coil and the output is the angular rotation of the arm, ϕ. The movement equation is:

$$J\frac{d^2\phi}{dt^2} = k' i_{rad}$$

This in Laplace notation:

$$\phi(s) = \frac{k'}{Js^2} i_{rad}(s)$$

This angular rotation is connected to the radial displacement by $x_{spot} = K_{arm}\phi$. The finite length of the rotating arm causes the factor K_{arm} to be non-linear. It can change by a factor 1.25 over the working area because of this. Due to other factors the process gain can vary in total by a factor 3 during operation. See section 3.5 for a more detailed discussion on the gain variation.

At "low" frequencies ($\leq 800Hz$) most components can be modelled as rigid bodies. However, at higher frequncies many flexible modes are present in the system, for instance bending modes of the rotating arm and torsional modes of the mounting plate. These flexible modes manifest themselves as resonance peaks in the frequency response.

The focus actuator, which focuses the laser spot on the disc, consists of a coil wound around the objective lens together with a permanent magnet. The objective lens is suspended by two leafsprings. The system therefore behaves as a mass-spring system. Its resonance frequency lies around $40Hz$.

Since the dynamic interaction between the two loops is relatively low, the two systems can be treated independently for control design. In current systems the two loops are single-input-single-output and are controlled by simple PID-controllers. The radial loop has a bandwidth of $500Hz$, while the focus loop has a bandwidth of $800Hz$.

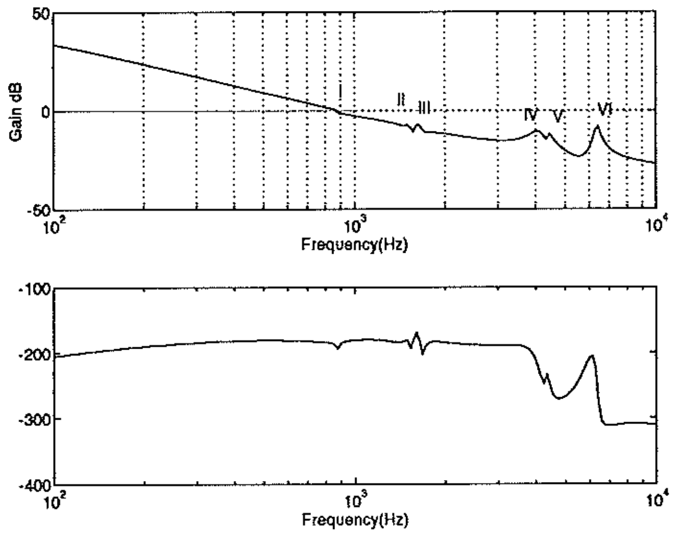

Figure 3.2 Bode diagram of 24th-order model of radial servo drive.

In this report the attention will be focused on the radial servo system. The focus control loop has smaller dynamical variations and is also less critical.

3.2 Model of the system

This section will provide a model of the radial servo system. Subsequently a block diagram of the system will be shown to clarify all the signals and transfer functions.

The actuator of the radial drive thus consists of an input which is the current to the coil and an output which is the tracking error. When one writes the movement equations one gets :

$$J\frac{d^2\phi}{dt^2} = k' i_{rad}$$

$$\phi(s) = \frac{k'}{Js^2} i_{rad}(s)$$

A simple model of the transfer function from i_{rad} to ϕ is thus a double integrator. This model will be used in the preliminary calculations to design the fixed controller. The factor k'/J can be determined by studying identified models of the CD-player and will be called K_p from now on.

At frequencies above $800Hz$ the frequency response of the actual system deviates from the simple model of the double integrator. The frequency response of a 24th order identified model of the radial drive [3] (figure 3.2) shows resonance peaks at $0.8, 1.6, 4.3$ and $6.5kHz$. These peaks are caused by flexible and torsional deformations of the mounting plate, radial arm, CD and turn-table motor together. The damping and the nominal frequencies of these peaks can vary because of production tolerances. One has to take these peaks into account when raising the bandwidth to $1kHz$ or higher. The reason for this can be seen best in a Nyquist curve of this 24th order model (figure 3.3). The various peaks in the Bode diagram appear as loops in the Nyquist curve. The numbered loops in the Nyquist curve correspond with the numbered peaks in the Bode diagram.

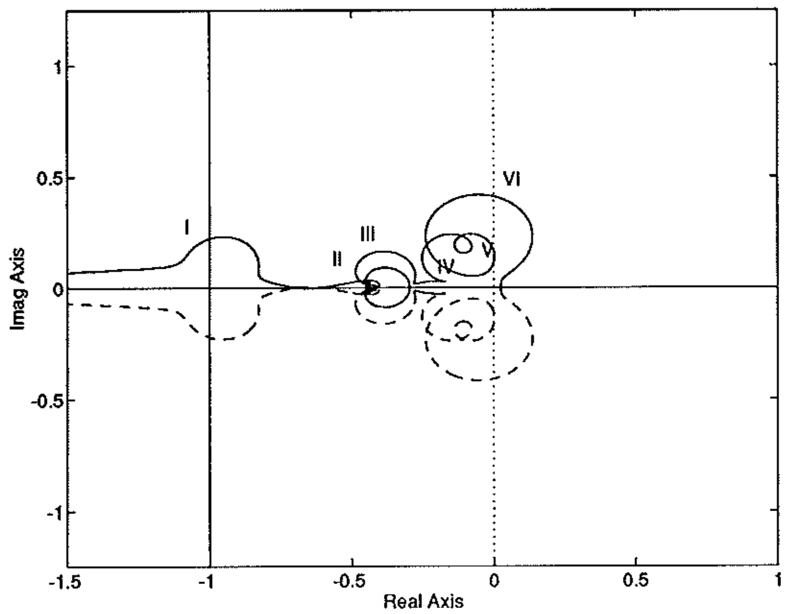

Figure 3.3 Nyquist diagram of 24th-order model of radial servo drive.

The transfer function for the radial actuator has thus been given. This function relates the input i_{rad} to the output e_{rad}. This transfer function incorporates the function that relates the angular rotation, ϕ, to the spot displacement x_{spot}; K_{arm}. Because of the rotating arm and other factors this function will be non-linear.

In figure 3.4 the blockdiagram of the system is shown. The process is represented by $G(s)$. This relates the input, the current to the coil, to the output, the angular rotation ϕ. K_{arm} relates the angular rotation to the actual spot position x_{spot}. The difference between the actual spot position and the reference signal, which is denoted by ϵ_{rad}, is measured by the optical pickup G_{opt} which produces the error signal e_{rad}. This signal can be used as an input to the controller or as a signal for identification. When the control signal i_{rad} is produced an additional signal r can be added to this. Usual practice is to insert a sinusoidal signal at this point, which is used to calculate the process gain (for the case the process gain is controlled using the Wobble method [1]).

Disturbances can enter the system at several points. For a detailed description on disturbances see sections 3.3 and 4.3.

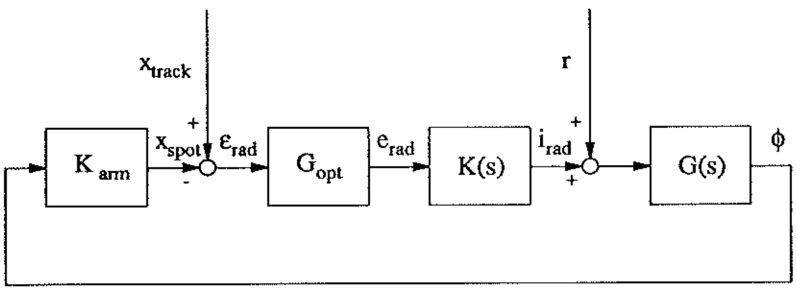

Figure 3.4 Blockdiagram of radial servo loop

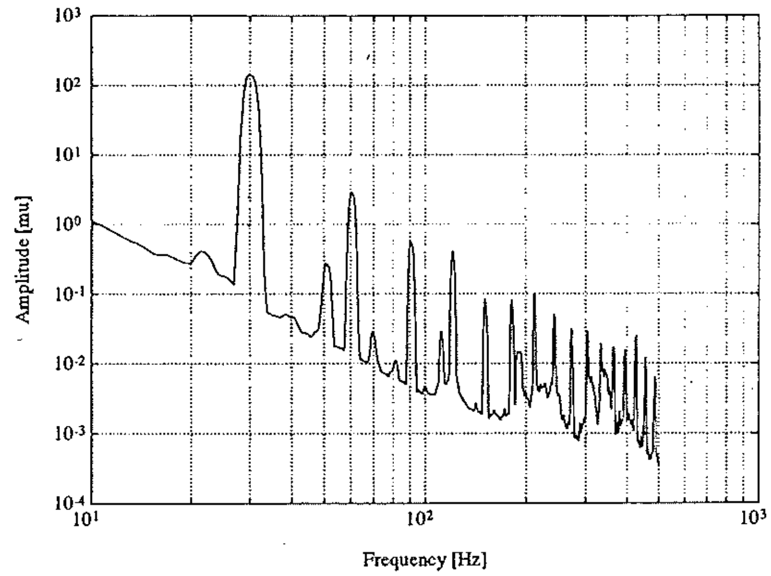

Figure 3.5 Radial tack spectrum of a worst case disc rotating at $30Hz$

3.3 Disturbances

This section will cover the nature and character of the disturbances working on the system. In figure 3.5 one can find the radial track spectrum of a worst case disc rotating at $30Hz$. A worst case disc is a disc that has the highest eccentricity possible that can pass disc specifications.

¿From figure 3.5 it can be seen that the most important track disturbance is the peak at the rotational frequency. This peak is caused by the eccentricity of the disc. At higher harmonics of the rotational frequency other peaks can be found which are also connected to the disc eccentricity. The amplitude of these higher harmonic disturbances is smaller. Because the rotational frequency of the disc is not constant during operation, the location of these peaks can vary. The peaks will thus shift as the rotational frequency changes.

Other disturbances working on the system at low frequencies are those caused by external shocks. These can be classified in disturbances under "jogging" conditions ($\approx 5Hz$) and shocks that result from, for example putting the player on a table (between 10 and $200Hz$). The two classes of disturbances, those caused by disc eccentricity and those caused by external shocks, thus mainly have effect for frequencies lower than $300Hz$.

Then there are disturbances that have effects over a wider frequency band. Their amplitude and effect is less than those already described. These disturbances can be classified into two categories. One are the disturbances caused by track irregularities, dust, scratches and noise generated by the pit structure. The other category is the usual disturbance component: measurement noise.

Because the effect of the lowfrequent disturbances is much higher than that of the other classes of disturbances, the term disturbance rejection will refer to the attenuation of the low frequency disturbance components.

3.4 Specifications

This section will cover the various specifications which the controlled system must satisfy.

The required bandwidth in current CD-systems of the radial servo system is $500Hz$ [4]. This value is chosen based upon conflicting factors: suppression of mechanical shocks affecting the player, achievement of sufficient disturbance attenuation, playability of discs containing faults, and power consumption. There is no official bandwidth specification for systems with higher performance requirements. In this research it is the aim to push up the bandwidth as far as possible.

The maximum allowable radial tracking error is $0.1\mu m$. To see how high the loop gain must be to get sufficient disturbance rejection, the various low frequent disturbance components must be examined. From figure 3.5 it can be seen that the most important disturbance is found at the rotational frequency. For a worst case disc the highest value of the amplitude of the sinusoidal disturbance signals is $100\mu m$. This peak can be found at the rotational frequency. The loop gain must thus be at least 1000 ($= 60dB$) to achieve the allowable tracking error. For the peaks at higher harmonics a similar calculation can be made. For the case that the pit size is decreased, which is desirable for data storage units, the specifications for the allowable tracking error will be tightened even more.

3.5 Gain variation

During operation the process gain is not constant. This is because of various reasons. The first reason is the fact that the pick-up unit rotates around a fixed point with a finite arm length. The movement of the laser spot is therefore not always perpendicular to the tracks. As can be seen from figure 3.6 it can vary because of this by a factor 1.25. Another cause of process gain variation is the use of production tolerances. The final reason for process gain variation can be found in the transfer function K_{opt}. The gain variation here is caused

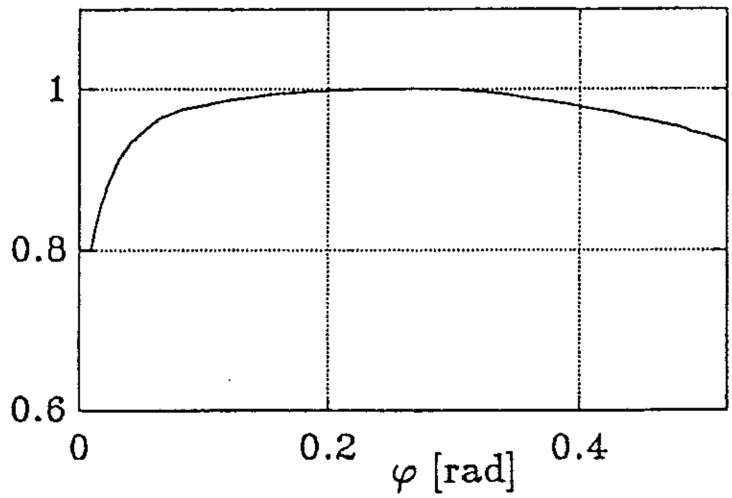

Figure 3.6 Gain variation in the radial actuator due to the finite arm length

by varying shape, depth and slope of the pits on the disc and variations in the optical quality of the transparent layers on the disc and lenses. These factors taken together yield a process gain that can vary by a factor 3.

The effect of the gain variation in the process is an increase or decrease of the bandwidth of the controlled process. The need for gain variation compensation is apparent. Without it the controlled system would have a very low performance, since the designed controller would be designed for the highest value of the process gain.

In earlier Philips CD-players this problem of varying process gain was solved using the technique of *Automatic Gain Control* (AGC). With this method a sine wave was injected into the system as an addition to the control signal (signal r in figure 3.4). With this signal it was possible to identify the actual bandwidth of the system and apply compensation if needed.

A different solution to this problem was developed by Draijer using an adaptive technique [4]. Using the adaptive scheme of the self tuning regulator the process gain was identified after which an inversion of the identified gain was inserted in the loop to achieve the required compensation. This adaptive controller works without any external excitation, thereby avoiding any unnecessary deviations of the laser spot from the center of the track.

3.6 High frequency dynamical variation

When raising the bandwidth of the system the problem of the effect of resonance peaks showing up prominently in the frequency response arises. The first one lies around $800Hz$, the other lie at higher frequencies. As already said, pushing up the bandwidth of the system means designing a controller that can deal with the high frequency peaks.

The real problem is not that they exist but that they vary in shape and place from player to player. The reason for this is the use of production tolerances during production of the various CD-player parts. Because of these applied tolerances the different parts of the system are not identical, causing behavioural differences in the final products. These varying behaviours are most prominent in the changing frequency and shape of the resonance peaks. How much the peaks can change and which peaks change most is not yet known. When designing a robust controller for this problem it was assumed that the nominal frequencies of the resonance peaks were uncertain with 2.5% [5].

4. Control design

When designing a controller for the current CD-player we have to take the following points into account that are characteristic for this system:

- The process is critically unstable for proportional feedback. It behaves approximately as a double integrator in open loop.
- We need a high lowfrequency disturbance attenuation. The system has large disturbances around the rotational frequency.
- The bandwidth should be pushed up as high as possible.
- There are resonance peaks at high frequencies.
- The system has a varying process gain during operation.
- Not every CD-player behaves the same. This is caused by production tolerances. The effect this has can be seen best in shifting high-frequency resonance peaks.

Of course there are already controllers being used in ordinary CD-players. However there is a need to improve the controllers. One would like to see an improvement of the disturbance rejecting qualities and also increase the rotational frequencies of certain CD-player types (CD-rom).

A simple way of achieving this improvement is to push up the bandwidth of the system. The lowfrequency disturbance behaviour improves, and the rotational frequency can be increased. As was seen in chapter 3 problems can arise when the desired bandwidth is placed too high. The effect of high frequency resonance peaks in the closed loop response can get too prominent. To get a satisfactorily response for high bandwidths one then has to deal with these peaks. The fact that these peaks can vary in shape and location from player to player causes extra problems in designing a controller for this system.

The solution to this is in the form of a flexible controller. It should have the flexibility to handle the varying resonance peaks. With an adaptive scheme it is possible to design a flexible controller that can handle the problem of varying system dynamics on-line. That means that a controller designed with adaptive characteristics can compensate the resonance peaks even if they shift or change shape during operation.

The controller is constantly tuned and retuned to the differing situation. This means that the controller will be more complex and bigger than a controller that is tuned once and is fixed after tuning.

4.1 General approach

The adaptive scheme will be built into every CD-player controller. This means that the controller will be able to adapt to varying dynamics during operation.

The controller will consist of two "parts". One is the fixed part, or rather the part that is non-flexible which is the same for every CD-player. It is designed using a low order model of the radial drive and does not take varying dynamics into account. The order is propably very low, around 3rd or 4th order. With this fixed part the first three points named in the introduction of this chapter are taken care of. The bandwidth of the controlled system will be placed as high as possible. The fixed controller will be designed keeping in mind that the most undesirable frequency peak will be cancelled by the adaptive controller.

The second part is the adaptive controller. This part will compensate the effects of one resonance peak as well as the phenomena of the varying process gain. These two operations can easily be taken together into one adaptive process. The method to compensate the peak as well as the varying process gain will be discussed in section 4.3.

An alternative to reduce the effect of the disturbances caused by eccentricity could have been one using gain scheduling. This thought arises because the disturbances caused by the eccentricity can be found at very specific locations. These locations are closely connected to the rotational frequency and shift as the rotational frequency changes. If the rotational frequency is measurable then gain scheduling could be used to let the controller adapt itself to the shifting disturbance peaks. Unfortunately, the rotational frequency is not known.

In section 4.2 the design of the fixed part of the controller is discussed. Section 4.3 then gives the design of the flexible part of the controller.

4.2 Design of fixed controller

With the fixed controller the first three points named in the section 4.1 of this chapter will be taken care of. This means that after the implementation of this fixed controller in the loop the system will be stabilized, it will heve enough lowfrequent disturbance attenuation and it will have the required bandwidth.

The stabilization of the system can be achieved by including a phase lead network in the controller. This can be done quite easily, only the zero and the pole of the network need to be chosen.

Sufficient disturbance attenuation can be achieved by adding a single or double integrator, or perhaps an inverted notch filter with its resonance at the maximal rotational frequency. Experiments showed that the use of the inverted notch filter provided a large loop gain at low frequencies, but also caused low damping for the closed loop system because of the phase lag it introduced around the bandwidth. The simple integrator, which showed less phase lag problems, also evidenced enough gain to handle the disturbances at low frequencies. This option was therefore chosen.

As mentioned before the required controller will be designed using the technique of pole-placement design. With this technique the desired closed loop response as well as the wanted controller structure can easily be specified.

The controller will be designed in continuous time using a simple double integrator as the process model. The controller will then be tested on a 24th order model of the system [3]. The simple process model is:

$$\frac{B}{A} = \frac{K_p}{s^2}$$

With these process polynomials all process zeros will be cancelled (there are no process zeros) so we can split the B-polynomial in the following way:

$$B^+ = 1$$

$$B^- = B$$

The diophantine equation then changes to:

$$AR + BS = A_m A_o$$

The desired closed loop system must now be specified to obtain the needed A_m-polynomial. To achieve a controller with the lowest degree $degA_m = degA$ should be chosen. The desired closed loop system is thus chosen of 2nd order. The structure of this second order model is:

$$\frac{B_m}{A_m} = \frac{\omega_{band}}{s^2 + 2\zeta\omega_{band}s + \omega_{band}^2}$$

The parameters ω_{band} and ζ are free to choose. As already mentioned ω_{band} is placed as high as possible and ζ is chosen to be 0.7.

The controller we can implement looks like:

$$Ru(t) = -Sy(t)$$

The T-polynomial present in the general controller is missing for this case. This is because the reference signal is not measurable.

Before the diophantine equation can be written out the order of the controller-polynomials must be specified. To contain a low-frequent integrator, a phase lead network and rolloff the R-polynomial must be at least of 3rd order. The S-polynomial must be of one order less to achieve the desired rolloff; S is thus of second order. This means that the A_o-polynomial is of 3rd order.

The Diophantine equation that delivers the controller with the specification named above then looks like:

$$s^2(s^2 + r_1 s + s_2)s + K_p(s_0 s^2 + s_1 s + s_2) =$$

$$(s^2 + 2\zeta\omega s + \omega^2)(s^3 + a_{s1}s^2 + a_{s2}s + a_{s3})$$

with:

$$\begin{cases} a_{s1} = a_1 + a_2 + a_3 \\ a_{s2} = a_1 a_2 + a_1 a_3 + a_1 + a_2 a_3 \\ a_{s3} = a_1 a_2 a_3 \end{cases}$$

As can be seen, the lowfrequent integrator is specified by splitting up the polynomial R into an unknown second order part and a known part, s. Results show that if the following observer poles are chosen the desired closed-loop bandwidth will be achieved:

$$\begin{cases} a_1 = 150 \\ a_2 = 150 \\ a_3 = 25 \end{cases}$$

This is if the desired bandwidth is $500Hz$. For higher bandwidths the observer poles should be adjusted accordingly.

The diophantine equation can now be solved to obtain the controller parameters. The solution is:

$$\begin{cases} r1 = a_{s1} + 2\zeta\omega \\ r2 = a_{s2} + a_{s1}2\zeta\omega + \omega^2 \\ s0 = (a_{s3} + 2\zeta\omega a_{s2} + \omega a_{s1})/K_p \\ s1 = (2\zeta\omega a_{s3} + \omega^2 a_{s2})/K_p \\ s2 = (\omega^2 a_{s3})/K_p \end{cases}$$

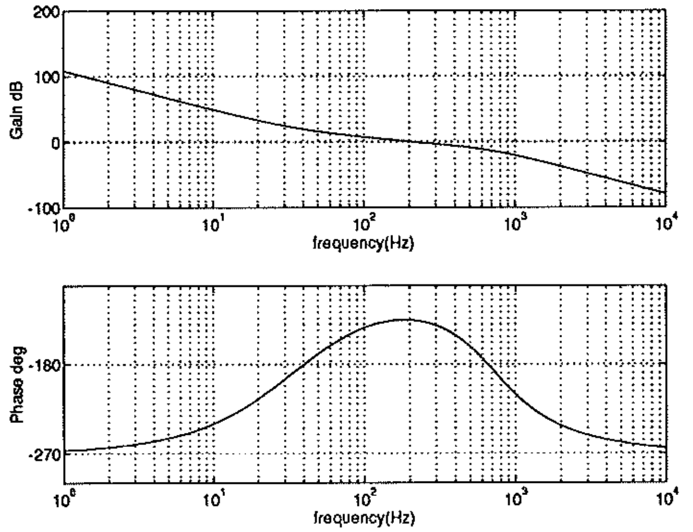

Figure 4.1 Bode diagram of loop gain of second order model controlled with fixed controller with desired bandwidth at $500Hz$

Having designed the controller it is now to be seen if it fits the specifications. A Bode diagram and a Nyquist curve of the loop transfer function can be found in figures 4.1 and 4.2 for a desired bandwidth of $500Hz$. In these figures a simple double integrator is used as the process model. With this controller the loop gain at $8Hz$ is $52dB$; not enough to satisfy the specification of $60dB$. However there is enough space to push up the bandwidth and improve this gain.

With this model of the double integrator the bandwidth can be pushed up as far as one wants. However, if the designed controller is used with the high order model, problems arise connected to the presence of the resonance peaks. The highest value for the desired bandwidth, which still gives a reasonably

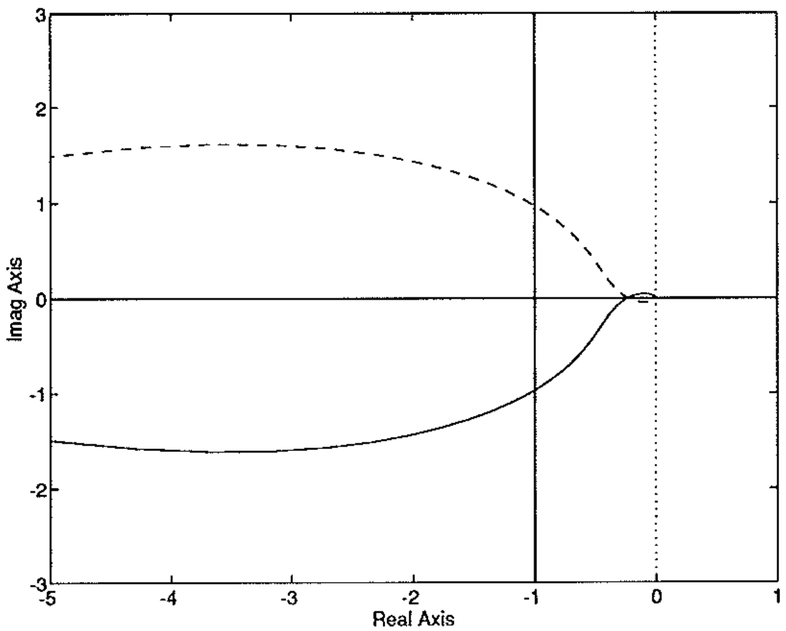

Figure 4.2 Nyquist diagram of second order model controlled with fixed controller with desired bandwidth at $500Hz$

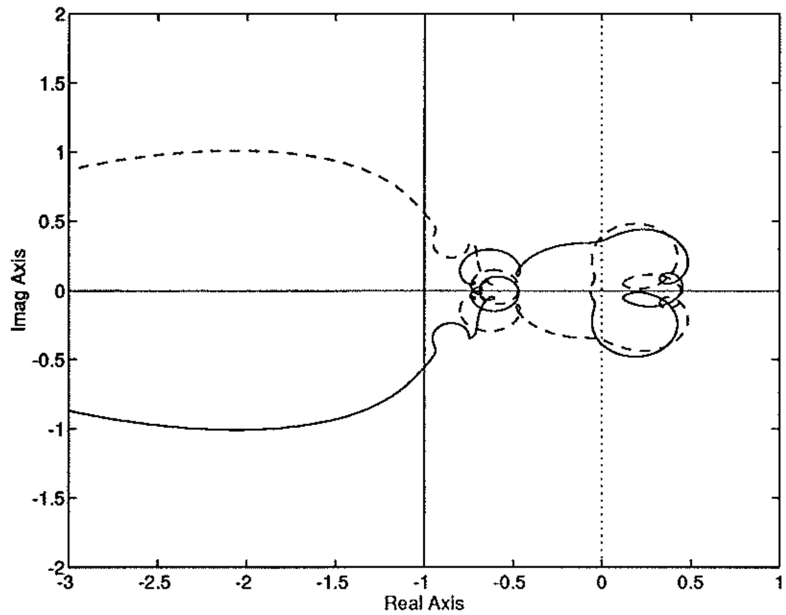

Figure 4.3 Nyquist diagram of 24th order model controlled with fixed part of controller with desired bandwidth at $1900Hz$

damped system, is around $1900Hz$. In figure 4.3 one can see a Nyquist curve of the controlled 24th order model using the controller designed for the desired bandwidth of $1900Hz$. Cleary can be seen that the resonance peaks cause loops in the nyquist curve that run close to the -1 point, thus restricting the value of the desired bandwidth. For the controller designed for a desired bandwidth of $1900Hz$ the loop gain at the rotational frequency of $8Hz$ is $86dB$, thus amply satisfying the disturbance rejection specification.

In figure 4.4 a stepresponse of the designed system is shown. The system is the 24th order model controlled with the fixed controller designed for a bandwidth of $1900Hz$. The overshoot of this system is clearly too high. Lowering the desired bandwidth will lower the overshoot. However, this lowering of the bandwidth will also diminish the effect of the resonance peaks on the systems behaviour. To achieve clear results for the adaptive controller later on, the desired bandwidth of the fixed controller will therefore not be lowered.

The question now is which peak is best to cancel. Figure 4.3 shows that the loops at $800Hz$ and at $1600Hz$ have approximately the same amount of damping for the designed fixed controller. Because it is not yet known which of these two peaks shows the most variation in practice, it is hard to say which peak is best to cancel. It is chosen to cancel the peak at $800Hz$.

4.3 Design of adaptive scheme

The flexible part of the controller will try to compensate for or weaken the effect of the harmful resonance peak at $800Hz$ as well as compensate for the varying process gain. The technique that is used is the method of the self tuning regulator. This section treats the design and implementation of such a regulator.

First the required peak and the process gain have to be identified using the signals e_{rad} and i_{rad}. In order to do this information about the peak in these two signals must first be isolated from the rest of the information in the

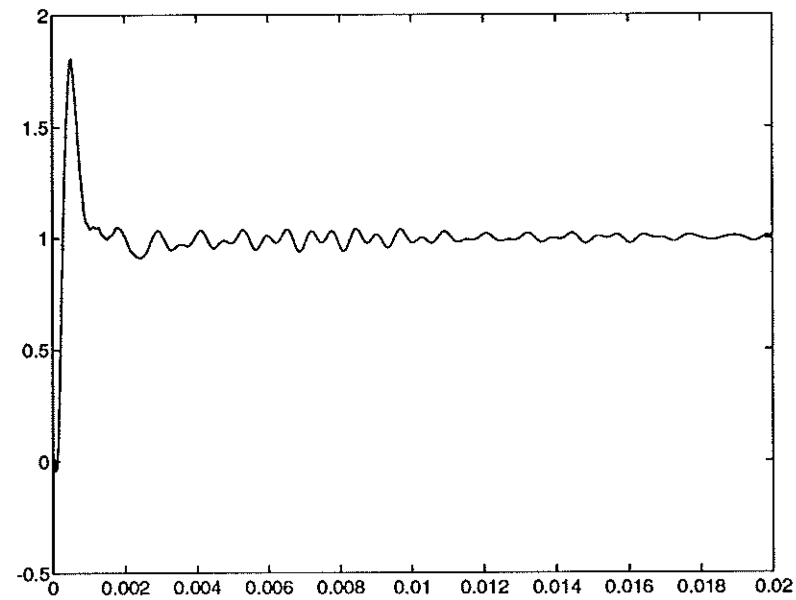

Figure 4.4 Step response of 24th order model controlled with fixed part of controller with desired bandwidth at $1900Hz$

signals. This is done using bandfilters and an *a priori* filter. Using these filters information about the process gain is also passed through. In the estimation stage the required peak and the process gain are then identified simultaneously. The model of the peak will then be used for the control design of the flexible part of the controller. The design stage will consist of simply inverting the identified model, and using this inverted model as extra controller in the loop.

Prefiltering

When performing system identification of a certain process, the ideal signals to work with are the process input and the process output. For this case these signals are i_{rad} and x'_{spot} ($x'_{..}$ is the signal $x_{..}$ fed through K_{opt}). The signal i_{rad} is known, the signal x'_{spot} is not. Only the error signal e_{rad}, which is the difference between x'_{spot} and x'_{track}, is a measurable signal. To find out if it is possible to design an adaptive system for this problem two things must be investigated. First of all a scheme must be found to identify only the required peak and process gain. Secondly, it must be investigated if this is possible using only the signals i_{rad} and e_{rad}.

The goal of the identification stage is to identify the process gain as well as the desired peak (the desired peak is the one at $800Hz$). These specific dynamics can be identified in an estimation stage by filtering the signals with bandfilters in combination with an *a priori* filter. The a priori knowledge of the second order process dynamics are used to simplify the estimator by filtering the control signal i_{rad}. The bandfilters are used to concentrate the estimation on the frequency band of the resonance peak. Both the control signal and the radial error signal e_{rad} have to be filtered by the bandfilter in order to maintain the same input-output behaviour of the filtered signals. With the use of this filtering stage it is thus possible to only identify the desired dynamics.

Now the problem must be handled of performing the identification using only the available signals. The two signals e_{rad} and i_{rad} must be examined carefully. In section 3.3 it is shown that the reference signal x_{track} has a large influence on the signal e_{rad} for frequencies lower than $300Hz$. If a high-pass filter is used to filter the signal e_{rad} it can minimize the effect of x_{track} on the parameter estimation by filtering out its most important components. In [4], in which a parameter estimation was designed to estimate the process gain of this system, this technique of filtering with a high-pass filter was used. The filter used in this research was:

$$H_f(s) = \frac{s^2}{(s-q)^2}$$

This filter gave the best estimation results if the break-off frequency was placed at the bandwidth of the system. In [6] it is proven that under the assumption of white noise disturbance, an optimal estimation for closed-loop conditions is obtained when the pre-filters used are equal to the sensitivity function. For the estimation of the process gain the ideal filter is thus a second order high-pass filter with the cut off frequency at the bandwidth, which is $1900Hz$.

For the estimation of the required resonance peak the filtering should be such that only the important frequency band is passed through. This can be achieved by using a bandfilter of the form:

$$H_f = \frac{s^2\omega_f^4}{(s^2 + 2\zeta_f\omega_f s + \omega_f^2)^2}$$

The resonance frequency ω_f is placed around $800Hz$. The filter is chosen to be 4th order for the same reason the high pass filter is chosen to be of 2nd order; to minimize the effect of the signal x_{track} on the estimation.

To combine the two estimations into one operation the two filters must be combined. A choice must be made when combining the two because the cut-off frequencies are not the same. The prefilter is chosen to be the ideal filter for the estimation of the resonance peak. This choice is made because the importance of the placing of the cut-off frequency is much greater for the case of the estimation of the resonance peak than it is for the estimation of the process gain. With this prefilter the peak as well as the process gain will be identified. It must however be kept in mind that the estimation of the process gain will not be ideal. The final prefilter thus looks like:

$$H_f = \frac{s^2\omega_f^4}{(s^2 + 2\zeta_f\omega_f s + \omega_f^2)^2}$$

in combination with the a priori filter:

$$H_{apriori} = \frac{K_p}{s^2}$$

with:

$$\begin{cases} K_p = 7e5 \\ \omega_f = 875Hz \\ \zeta_f = 0.2 \end{cases}$$

Estimation

The most important part of an STR is the parameter estimator. The method used to estimate the process parameters is the recursive least-squares estimation method with exponential forgetting [8]. The signals used for the estimation are the filtered e_{rad} and i_{rad} signals.

The estimation is recursive in order to save computation time. The computations are arranged in such a way that the results obtained at time $t-1$ can be used in order to get the estimates at time t. The strong point of the least squares method is the speed of response.

The least-squares estimate $\hat{\theta}$ satisfies the following recursive equation:

$$\hat{\theta}(t) = \hat{\theta}(t-1) + K(t)(y(t) - \phi^T(t)\hat{\theta}(t-1))$$

where:

$$K(t) = P(t)\phi(t) = P(t-1)\phi(t)(\lambda I + \phi^T(t)P(t-1)\phi(t))^{-1}$$
$$P(t) = (I - K(t)\phi^T(t))P(t-1)/\lambda$$

These equations have a strong intuitive appeal. The old parameter estimation is corrected with a term that is proportional to the difference between the measured value of $y(t)$ and the prediction of $y(t)$ based on the previous estimates of the parameters. $K(t)$ is a vector of weighting factors that tell how the correction and the previous estimate should be combined.

Because we are dealing with a system that is varying in time the estimation method must react quick during the complete operation period. There are two ways to achieve this. For systems that have abrupt parameter changes the best option is to reset the covariance matrix P periodically to αP, with α a large number. The other option is to introduce exponential forgetting in the form of the parameter λ $(0 \leq \lambda \leq 1)$. With this factor a time-varying weighting of the data is introduced. The most recent data is given unit weight, data that is n units old is weighted λ^n. A disadvantage of this method is that data is discounted even if there is no new information in the new data.

The required peak will be estimated with a 2^{nd} order model. This is justifiable because of the simple form of the peak at $800Hz$. It is also the lowest order that can identify a resonance peak of this form. The model will look like:

$$G_m(q^{-1}) = \frac{\theta_1 + \theta_2 q^{-1} + \theta_3 q^{-2}}{1 + \theta_4 q^{-1} + \theta_5 q^{-2}}$$

The five parameters θ_1 through θ_5 are estimated using the method discussed above..

Control design

Having obtained the various parameters of the required resonance peak and the process gain, a controller must be designed that can compensate the peak and handle the varying gain. This is done by simply inverting the found model which was found. A model of the form:

$$H_m(q^{-1}) = \frac{\theta_1 + \theta_2 q^{-1} + \theta_3 q^{-2}}{1 + \theta_4 q^{-1} + \theta_5 q^{-2}}$$

has an inverted model:

$$H_m(q^{-1}) = \frac{\theta_{c1} + \theta_{c2} q^{-1} + \theta_{c3} q^{-2}}{1 + \theta_{c4} q^{-1} + \theta_{c5} q^{-2}}$$

with:

$$\begin{cases} \theta_{c1} = 1/\theta_1 \\ \theta_{c2} = \theta_4/\theta_1 \\ \theta_{c3} = \theta_5/\theta_1 \\ \theta_{c4} = \theta_2/\theta_1 \\ \theta_{c5} = \theta_3/\theta_1 \end{cases}$$

This is now the extra controller part. With this construction the process gain is inverted and added to the controller and an inversion of the identified peak is added in the loop. For the used prefilters the identified process gain will be one if no correction is needed.

4.4 Summary

This section will summarize the controller design proposed as such, together with some issues connected to implementation.

The fixed part of the controller is designed using pole-placement design. It contains an integrator at low frequencies to improve disturbance rejection, phase lead to stabilize the system and high frequency rolloff to reduce the effect of high frequency dynamics. It is thus of 3rd order. Using this technique to design the controller the bandwidth can be pushed up to $1900Hz$ before high frequency resonance peaks start to have a large effect on the behaviour of the controlled system. The controller designed for a desired bandwidth of $1900Hz$ provides enough gain to satisfy the disturbance rejection specifications.

An adaptive scheme will compensate the effect of one resonance peak as well as the varying process gain. The designed adaptive controller is of very high complexity. For the prefiltering of the signals two bandfilters of 4th order in combination with an a priori filter of 2nd order are required. Further more a tunable 2nd order controller is needed. Finally the estimation and design stage of the adaptive scheme will take up a great amount of calculations. The complexity of this adaptive scheme exceeds the complexity of the adaptive scheme designed in [4] by a great deal. In the named research the designed adaptive scheme only compensated the varying process gain. This simple adaptive system already showed to have problems with the short sampling time with regard to the amount of calculations needed. The expectation is therefore that the proposed adaptive scheme, which also cancels the effect of one resonance peak, will pose too high requirements on the computational system.

5. Simulation results

The simulations for the proposed solution have been carried out using the program SIMNON. This program makes it possible to solve differential equations, difference equations and simulate dyamical systems that are composed of subsystems. In appendix A listings can be found of the SIMNON-code which were used for the simulations. This code has been changed at certain points to obtain the different simulation-instances. In the following sections a description of the simulation model will be given as well as a description of some of the simulations.

5.1 Simulation models

SIMNON showed some computational problems when using high order models of systems. Because of this, the model of the radial drive was lowered in order to get functioning simulations. In figure 5.1 a Nyquist plot is shown of the used 12th order model of the radial drive that is controlled by the designed fixed controller. The important resonance loops at 800, 1600 and 4000 are present in this low order model. They are, however, more prominent than in the 24th order model presented in section 3.2.

The used signal filters are the following:

- for the signal i_{rad} :

$$H_{f,i_{rad}} = \frac{K_p\omega_f^4}{(s^2 + 2\zeta_f\omega_f s + \omega_f^2)^2}$$

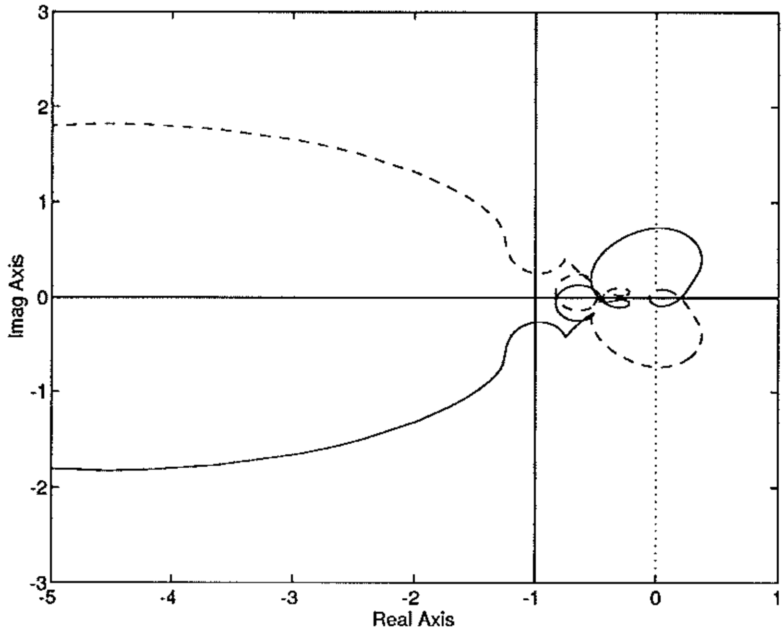

Figure 5.1 Nyquist diagram of system used for simulations : 12th order model controlled by designed fixed controller

- for the signal e_{rad} :

$$H_{f,e_{rad}} = \frac{s^2\omega_f^4}{(s^2 + 2\zeta_f\omega_f s + \omega_f^2)^2}$$

with :

$$\begin{cases} K_p = 7e5 \\ \omega_f = 875Hz \\ \zeta_f = 0.2 \end{cases}$$

The used estimation procedure is the recursive least squares estimation with forgetting factor. Good results were obtained using a forgetting factor of 0.99. The initial covariance matrix was a matrix with 1e18 on the diagonal. To ensure sufficient excitation for the parameter estimation a random signal was used as a reference signal.

5.2 Simulation 1 : Adaptation to resonance peak

In this simulation the adaptation of the adaptive controller to the unwanted resonance peak will be examined. This will be done by showing the system without and with adaptation.

The simulation will start with no adaptation at all. The initial value of the flexible part is a simple proportional gain of 1. In figure 5.2 a response of the system can be found. Until $t = 0.04$ the reference signal is a square wave. The system has a badly damped mode which is caused by the loop at $800Hz$. At time $t = 0.04$ the reference signal changes into a random signal to achieve sufficient excitation. At $t = 0.05$ the adaptation is initiated. The estimated parameters converge to their final value quickly. Because the effect of the adaptation can not be seen easily from the random signal response, the reference signal is changed back to the square wave. The oscillating mode has almost completely disappeared, leaving only oscillations in higher frequencies.

5.3 Simulation 2 : Adaptation to varying process gain

This simulation will show results of the adaptive system responding to a varying process gain. The system will have the adaptation running full-time while the gain of the process is varied with a factor $\sqrt{3}$ in both directions.

The simulation starts with the process gain at $\sqrt{3}$ of the normal value. The flexible part of the controller again starts as a proportional gain of 1. Every $0.04sec$ the gain is increased or decreased with a factor 3. The speed of response can be influenced by changing the forgetting factor λ. If the forgetting factor is set at 0.98 then the speed of response resembles the speed of response of Draijers [4] system during simulations. Simulation results can be found in figure 5.3.

5.4 Summary

Simulation 1 shows that the performance of the process controlled with the fixed controller improves greatly when the peak at $800Hz$ is compensated with

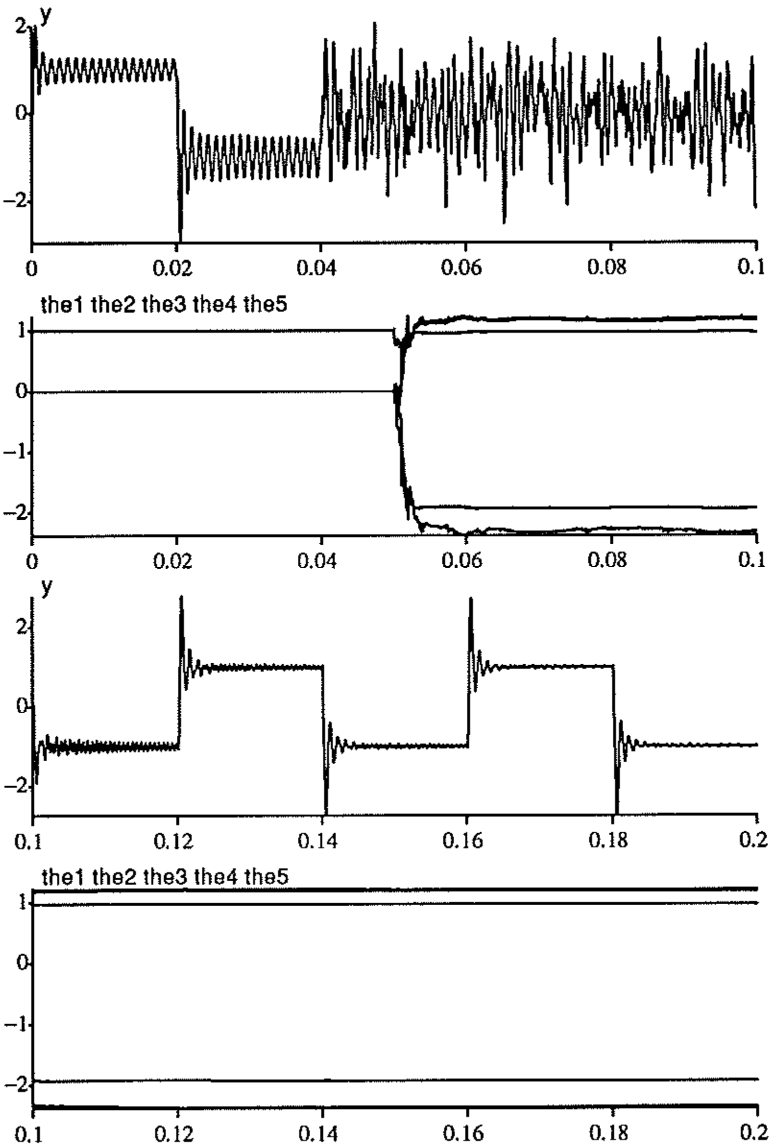

Figure 5.2 Results simulation 1 : Adaptation to resonance peak.

the designed adaptive scheme. The oscillation caused by this resonance peak disappears almost completely. Only higher frequency oscillations remain in the response of the system. Simulation 2 shows that the adaptive system also handles the process gain variation quite well. Gain variations that would normally push the system into instability are compensated for quite fast. However, the response of the system does react quite heavily to these gain variations. This is mainly the case when the process gain is increased.

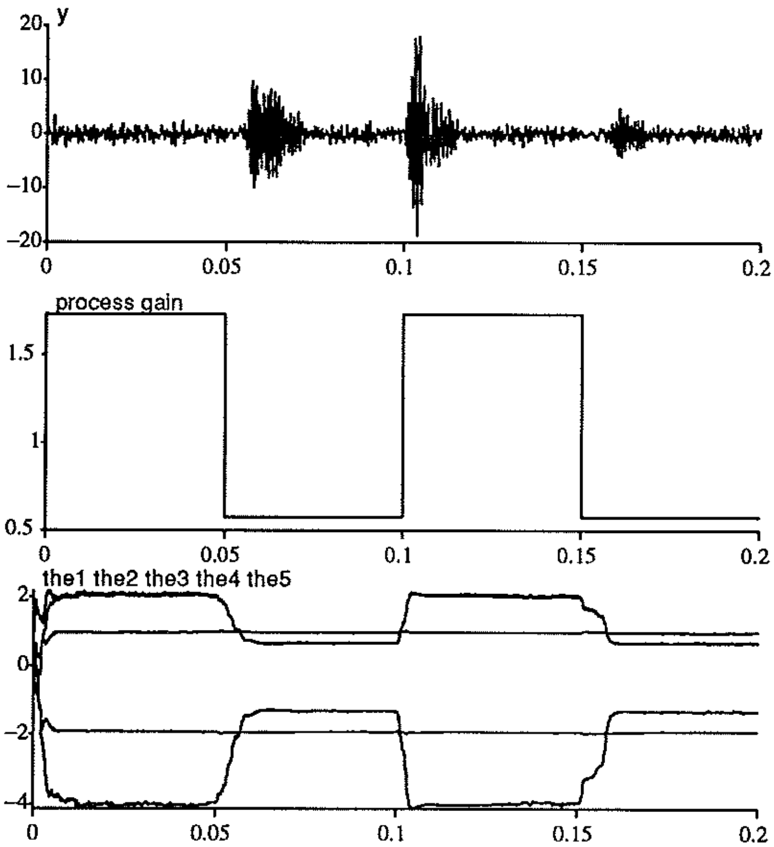

Figure 5.3 Results simulation 2 : Adaptation to varying process gain.

6. Evaluation

In this report, the use of adaptive control techniques has been investigated to realize a flexible controller for the laser spot positioning of the rotating arm Compact Disc mechanism CDM-9.

In section 6.1 the conclusions are stated that arise from this research. These conclusions are complemented with recommendations in section 6.2.

6.1 Conclusions

The conclusions of this research are :

- Using adaptive techniques it is possible to design a controller that is flexible enough to adapt to varying resonance peaks as well as a varying process gain.
- The adaptive scheme that was most suitable to handle varying high frequency dynamics was the technique of the self tuning regulator. This is because of its low-complexity and the ease with which it tackles complex problems, and because it leaves the user ample freedom to choose the various subfunctions of the method: estimation, controller design etc.
- The approach taken to compensate for high frequency dynamics was as follows:
  - Identify the most harmful resonance peak in the frequency response. This is done using filters for the controller input and output signals which isolate the required information.
  - An inversion of the identified model of the peak is used as a extra controller in the loop.
- The adaptive scheme needed to compensate for the varying process gain had already been developed. However, because the process gain is automatically controlled using the technique described above this has become obsolete.
- Using this technique to handle the problem the resulting adaptive controller is of very high complexity.
- The order of an adaptive system designed in this way is probably too high to handle the amount of calculations needed to update and identify all the needed states in one sample period.
- This technique of cancelling unwanted dynamics can be applied to more than one resonance peak. However, the order of the system will grow proportionally to the number of unwanted peaks.
- Simulations show the following results:
  - The adaptive system compensates the desired peak satisfactorily. The response is fast, and with suitably chosen signal filters it can compensate for peak shifts over the variation range.
  - The adaptive system also compensates for the varying process gain. Even when the process gain changes with the maximum value, which would normally push the system into instability, the system can compensate.

6.2 Recommendations

The conclusions made in this report are the basis for the following recommendations:

- A study must be made if it is necessary to compensate for varying high frequency resonance peaks during operation. Results already indicate that the high frequency dynamics vary from player to player but do not shift much during operation.
- Tests must be carried out to which extent the high frequency dynamics vary from player to player.
- Research must give further insight into which part of the high frequency dynamics can best be cancelled using flexible controller design techniques.
- It seems more practical and cost effective to compensate unwanted dynamics in the following way:
  - Split the controller into two parts. As was done in this report, make one part fixed and the other tunable.
  - Design the fixed part of the controller for the overall response keeping in mind that the unwanted high frequency dynamics will be taken care of by the tunable part of the controller.
  - Fit the design of the tunable part of the controller to the unwanted dynamics. These specific dynamics are identified at the end of the production line.

  This approach to design will lead to a controller that is much simpler than the controller based on adaptive techniques. However, it is no longer capable of retuning itself when dynamics change during operation. Because the dynamics do not change that much this will not be prove to be such a large problem.

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A. SIMNON code

This section will present the basic SIMNON-code used in the simulations. The total system is devided into various subsystems such as the estimation stage, the design stage, the process itself etc. SIMNON showed some calculation difficulties when handling high order systems (order higher than 6 or 7). Some subsystems are therefore devided into smaller parts. This is done, for instance, for the simulation model of the process which is a 12th order model. It is split into two blocks of 5th and 7th order.

The effect of roundoff errors is reduced by implementing the various models in SIMNON using the shift-controllable canonical form [9] instead of the direct form. There are other alternatives to implement the models which reduce the quantization even more, for instance the Parallel form or the Ladder form. Because the behaviour of the simulation models was good enough using the shift-controllable canonical form these were not used.

The code for the simulation models of the process are:

```
DISCRETE SYSTEM proc1
" simnon code for part of the process model
" 5th order model

state x5 x4 x3 x2 x1
new nx5 nx4 nx3 nx2 nx1
input ua
output utu
time t
tsamp ts

nx5= x4
nx4= x3
nx3= x2
nx2= x1
nx1= ap1*x1+ap2*x2+ap3*x3+ap4*x4+ap5*x5+ua
utu= cp1*x1+cp2*x2+cp3*x3+cp4*x4+cp5*x5
ts=t+h

" parameters of model1 -------------------

ap1= 3.665525818404090
ap2=-5.743621476581517
ap3= 4.922877129826090
ap4=-2.395776145056441
ap5= 5.509946734077790e-1

cp1= 4.329048777162978e-1
cp2=-1.316693931157771
cp3= 1.658335878662943
cp4=-1.011478209984283
cp5= 2.659234411562092e-1

h:0.00004
```

```
end
```

and:

```
DISCRETE SYSTEM proc2
" simnon code for part of the process model
" 7th order model

state x6 x5 x4 x3 x2 x1
state x7
new nx6 nx5 nx4 nx3 nx2 nx1
new nx7
input utu
output y
time t
tsamp ts

nx7= x6
nx6= x5
nx5= x4
nx4= x3
nx3= x2
nx2= x1
nx1= ap1*x1+ap2*x2+ap3*x3+ap4*x4+ap5*x5+ap6*x6+ap7*x7+utu
y  = cp1*x1+cp2*x2+cp3*x3+cp4*x4+cp5*x5+cp6*x6+cp7*x7
ts = t+h

" parameters of model2 ------------------

ap1= 4.509031028086261
ap2=-9.591272672557380
ap3= 1.276696961875243e1
ap4=-1.145673688715506e1
ap5= 6.879787425407230
ap6=-2.593842293081987
ap7= 4.860637805485117e-1

cp1= 4.036022575446045e-1
cp2=-1.526198407689271
cp3= 2.733337136189949
cp4=-3.023319349626671
cp5= 2.174853412955576
cp6=-9.590499328452382e-1
cp7= 2.067929479217002e-1

h:0.00004

end
```

The controller designed for a desired bandwidth of $1900Hz$ using pole-placement design is implemented as:

```
DISCRETE SYSTEM control

"Fixed part of controller

state x1 x2 x3
new nx1 nx2 nx3
input y
output u
time t
tsamp ts

r  = if ts>0.04 and ts<0.1 then norm(t) else sqw(t/0.04)
"r  = sqw(t/0.04)
erad=r-y
nx1= ar1*x1+ar2*x2+ar3*x3+erad
nx2= x1
nx3= x2
u  = cr1*x1+cr2*x2+cr3*x3

ts=t+h

" parameters of designed controller ---------------

ar1= 2.091554860193851
ar2=-1.467271412225689
ar3= 3.757165520318376e-1

cr1= 1.249213912935232
cr2=-2.405572814468136
cr3= 1.157910161654545

h:0.00004

end
```

The used filters :

```
DISCRETE SYSTEM bandfil

"bandfilter that filters the a priori filter output and
"the process ouput

input ufilu y
output ufil yfil
state xu1 xu2 xu3 xu4 xy1 xy2 xy3 xy4
new nxu1 nxu2 nxu3 nxu4 nxy1 nxy2 nxy3 nxy4
time t
tsamp ts

nxy4= xy3
nxy3= xy2
```

```
nxy2= xy1
nxy1= ay1*xy1+ay2*xy2+ay3*xy3+ay4*xy4+y
yfil= cy1*xy1+cy2*xy2+cy3*xy3+cy4*xy4

nxu4= xu3
nxu3= xu2
nxu2= xu1
nxu1= ay1*xu1+ay2*xu2+ay3*xu3+ay4*xu4+ufilu
ufil= cy1*xu1+cy2*xu2+cy3*xu3+cy4*xu4

ts=t+h

"parameters of bandfilter--------------------------

ay1= 3.819587955276077
ay2=-5.561220951969085
ay3= 3.655169809718722
ay4=-9.157608767237753e-1

cy1= 0
cy2= 1.506826887630504e-9
cy3=-3.013653775261007e-9
cy4= 1.506826887630504e-9

h=0.00004

end
```

and the a priori filter:

```
DISCRETE SYSTEM aprifil

"a priori filter that filters process input

input u
output ufilu
state xu1 xu2
new nxu1 nxu2
time t
tsamp ts

nxu2= xu1
nxu1= au1*xu1+au2*xu2+u
ufilu= cu1*xu1+cu2*xu2

ts=t+h

"parameters of a priori filter--------------------------

au1= 2
au2=-1
```

```
cu1= 2.4e-2
cu2= 2.4e-2

h=0.00004

end
```

The parameter estimation stage looks like:

```
DISCRETE SYSTEM ident

" estimation stage of adaptive controller

input ufil yfil                   "filtered process output & input
output the1 the2 the3 the4 the5
state yfil1 yfil2 ufil1 ufil2     "controller states
state th1 th2 th3 th4 th5         "parameter estimates
state p11 p12 p13 p14 p15         "covariance matrix
state     p22 p23 p24 p25
state         p33 p34 p35
state             p44 p45
state                 p55
new nyfil1 nyfil2 nufil1 nufil2
new nth1 nth2 nth3 nth4 nth5
new n11 n12 n13 n14 n15 n22 n23 n24 n25 n33 n34 n35 n44 n45 n55
time t
tsamp ts

"Computation of P*f and estimator gain k --------------------

pf1 = if t<instap then 0 else p11*f1+p12*f2+p13*f3+p14*f4+p15*f5
pf2 = if t<instap then 0 else p12*f1+p22*f2+p23*f3+p24*f4+p25*f5
pf3 = if t<instap then 0 else p13*f1+p23*f2+p33*f3+p34*f4+p35*f5
pf4 = if t<instap then 0 else p14*f1+p24*f2+p34*f3+p44*f4+p45*f5
pf5 = if t<instap then 0 else p15*f1+p25*f2+p35*f3+p45*f4+p55*f5
denom = lambda+f1*pf1+f2*pf2+f3*pf3+f4*pf4+f5*pf5
k1 = pf1/denom
k2 = pf2/denom
k3 = pf3/denom
k4 = pf4/denom
k5 = pf5/denom

"Update estimates and covariances ---------------------

ysch = f1*th1+f2*th2+f3*th3-f4*th4-f5*th5
eps  = if t<instap then 0 else yfil-ysch
"eps = yfil-ysch
nth1 = th1+k1*eps
nth2 = th2+k2*eps
nth3 = th3+k3*eps
nth4 = th4-k4*eps
nth5 = th5-k5*eps
```

```
n11 = if p22>po then po else (p11-pf1*k1)/lambda
n12 = (p12-pf1*k2)/lambda
n13 = (p13-pf1*k3)/lambda
n14 = (p14-pf1*k4)/lambda
n15 = (p15-pf1*k5)/lambda
n22 = if p33>po then po else (p22-pf2*k2)/lambda
n23 = (p23-pf2*k3)/lambda
n24 = (p24-pf2*k4)/lambda
n25 = (p25-pf2*k5)/lambda
n33 = if p44>po then po else (p33-pf3*k3)/lambda
n34 = (p34-pf3*k4)/lambda
n35 = (p35-pf3*k5)/lambda
n44 = if p11>po then po else (p44-pf4*k4)/lambda
n45 = (p45-pf4*k5)/lambda
n55 = if p55>po then po else (p55-pf5*k5)/lambda

"Form regression vector ------------------------------

f1 = ufil
f2 = ufil1
f3 = ufil2
f4 = yfil1
f5 = yfil2

"Update controller states ---------------------------

nyfil1 = yfil
nyfil2 = yfil1
nufil1 = ufil
nufil2 = ufil1

"Ranaming ------------------------------------------

the1=th1
the2=th2
the3=th3
the4=th4
the5=th5

"Update sampling time

ts = t+h

" Parameters ----------------------------------------

th1: 1
th2: 0
th3: 0
th4: 0
th5: 0
p11:1e18
p22:1e18
```

```
p33:1e18
p44:1e18
p55:1e18
po:1e18
lambda=if ts>0.1 then 1 else 0.99
instap:0.05
h:0.00004

end
```

The control design stage of the adaptive controller is:

```
DISCRETE SYSTEM adapt

" controller design of the adaptation part

state x3 x2 x1
new nx3 nx2 nx1
input u the1 the2 the3 the4 the5
output ua
time t
tsamp ts

nx3= x2
nx2= x1
nx1= -a1*x1-a2*x2-a3*x3+u
ua =  b3*x3+b2*x2+b1*x1
ts =t+h

k=60000

r1=1-exp(-k*h)
r2= -exp(-k*h)

a1 = (the2/the1)+r2
a2 = (r2*the2/the1)+the3/the1
a3 = r2*the3/the1
b1 = r1/the1
b2 = r1*the4/the1
b3 = r1*the5/the1

h:0.00004

end
```

The code that connects these various subsystems is:

```
CONNECTING SYSTEM connect

" connects various subsystems

time t
```

```
y[control]=y[proc2]
utu[proc2]=utu[proc1]
ua[proc1]=ua[adapt]
u[adapt]=u[control]

y[bandfil]=y[proc2]
u[aprifil]=ua[adapt]
ufilu[bandfil]=ufilu[aprifil]

yfil[ident]=yfil[bandfil]
ufil[ident]=ufil[bandfil]

the1[adapt]=the1[ident]
the2[adapt]=the2[ident]
the3[adapt]=the3[ident]
the4[adapt]=the4[ident]
the5[adapt]=the5[ident]

end
```

A macro that produces a simulation is:

```
MACRO simumac

"macro to obtain a simulation

SYST proc1 proc2 control bandfil aprifil ident adapt connect

store y[proc2] the1[ident] the2[ident] the3[ident] the4[ident]
store the5[ident] -add
split 2 1
simu 0 0.2
ashow 0 0.1 y
text 'y'
ashow 0 0.1 the1 the2 the3 the4 the5
text 'the1 the2 the3 the4 the5'

end
```