

Control Systems

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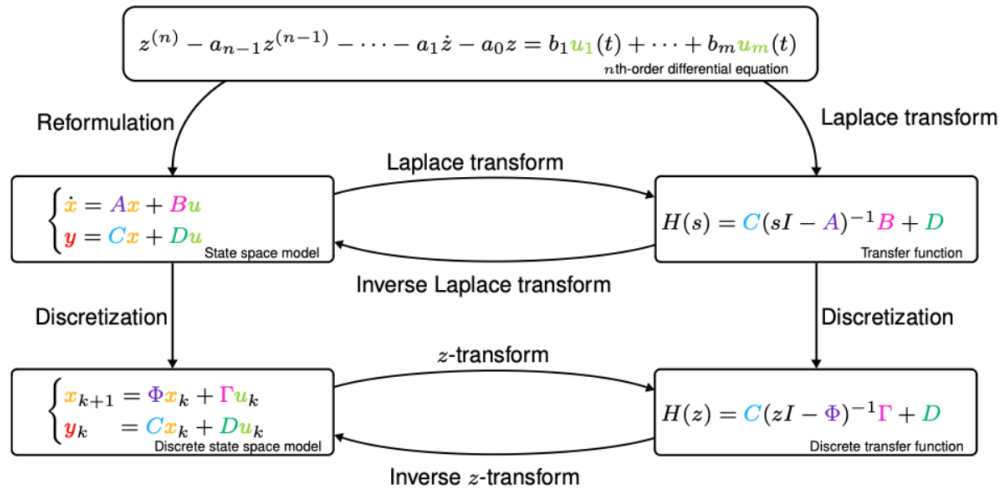
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1 Linear Time Invariant Systems

Overview



1.1 Time-Domain models

Linear Map

The map $f : \mathbb{R}^n \rightarrow \mathbb{R}^m$ is said to be linear if for any $x, y \in \mathbb{R}^n$ and $\alpha \in \mathbb{R}$, the following conditions hold

$$f(x + y) = f(x) + f(y) \quad \text{Super position}$$

$$f(ax) = \alpha f(x) \quad \text{Homogeneity}$$

The function has to go through (0,0) in 2D for it to be linear due to homogeneity.

Time-Invariant System

Let $\sigma : \mathbb{R} \times \mathbb{R}^m \rightarrow \mathbb{R}^p$ define the input-output behavior of a system model Σ . The system Σ is time-invariant if for any input signal $u : \mathbb{R} \rightarrow \mathbb{R}^m$ and any delay $\tau \in \mathbb{R}$ the following relation holds:

$$y(t - \tau) = \sigma(t, u(t - \tau))$$

for all times $t \in \mathbb{R}$, where y denotes the output signal of the system.

The importance is that the system does not change its behavior due to time. This can be seen as a canon firing at 8am it will not fire different compared to if you do the same at 5pm.

Two types of linear time-domain models.

Continous-time state space models (based on differential equations)

$$\dot{x} = Ax + Bu$$

$$y = Cx + Du$$

- $x \in \mathbb{R}^n$ is state. e.g. position or velocity
- $u \in \mathbb{R}^m$ is input
- $y \in \mathbb{R}^p$ is output
- $A \in \mathbb{R}^{n \times n}$ is system matrix

- $B \in \mathbb{R}^{n \times m}$ is input matrix
- $C \in \mathbb{R}^{p \times n}$ is output matrix
- $D \in \mathbb{R}^{p \times m}$ is the direct feedthrough matrix

Discrete-time state space models (based on difference equations)

$$x_{k+1} = \Phi x_k + \Gamma u_k$$

$$y_k = C x_k + D u_k$$

$$\Phi = e^{AT}$$

$$\Gamma = \int_0^T e^{A(T-\tau)} B d\tau$$

Matrix exponential can be computed with `expm` in Matlab.

1.2 Frequency-Domain models

Transfer function:

$$G(s) = \frac{Q(s)}{P(s)}$$

where $Q(s)$ and $P(s)$ are polynomials in s .

- The roots of $P(s)$ are called the **poles** of $G(s)$
- The roots of $Q(s)$ are called the **zeros** of $G(s)$

1.2.1 State space to transfer function

Taking Laplace tranforms of the system and assuming $x_0 = 0$:

$$\dot{x}(t) = Ax(t) + Bu(t)$$

$$y(t) = Cx(t) + Du(t)$$

yields:

$$sX(s) = AX(s) + BU(s)$$

$$Y(s) = CX(s) + DU(s)$$

Which can be rewritten as (I is the identity matrix):

$$X(s) = (sI - A)^{-1} BU(s)$$

$$Y(s) = \left(C (sI - A)^{-1} B + D \right) U(s)$$

where

$$Y(s) = G(s)U(s)$$

$$G(s) = C (sI - A)^{-1} B + D$$

1.2.2 Transfer function to state space

$$G(s) = \frac{1}{s^2 + a_1s + a_2}$$

$$y(s) = \frac{1}{s^2 + a_1s + a_2}u(s)$$

$$s^2Y(s) + a_1sY(s) + a_2Y(s) = U(s)$$

Inverse Laplace transform:

$$\ddot{y}(t) + a_1\dot{y}(t) + a_2y(t) = u(t)$$

Now $x_1 = y(t)$ and $x_2 = \dot{y}(t)$:

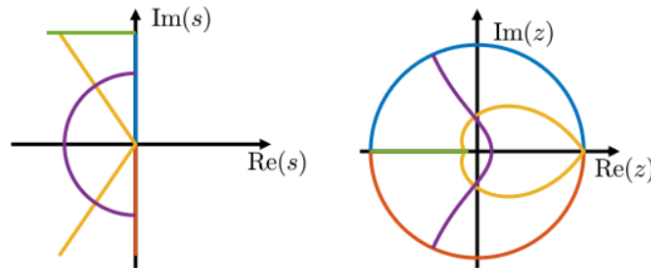
$$\begin{bmatrix} \dot{x}_1 \\ \dot{x}_2 \end{bmatrix} = \begin{bmatrix} 0 & 1 \\ -a_2 & -a_1 \end{bmatrix} \begin{bmatrix} x_1 \\ x_2 \end{bmatrix} + \begin{bmatrix} 0 \\ 1 \end{bmatrix} u$$

$$y = \begin{bmatrix} 1 & 0 \end{bmatrix} \begin{bmatrix} x_1 \\ x_2 \end{bmatrix} + (0)u$$

1.2.3 Discrete-time transfer function

Discretization from s -domain to z -domain can be done using:

- Matched z-transform
- Bilinear z-transform
- Impulse invariance z-transform



1.3 Examples

2 Stability and Performance Analysis

2.1 Basic System Classes

2.1.1 First Order Systems

State-space representation of first order system:

$$\begin{aligned}\dot{x} &= -\frac{1}{\tau}x + \frac{k}{\tau}u \\ y &= x\end{aligned}$$

A first-order system has one pole and is described by:

$$H(s) = \frac{k}{\tau s + 1}$$

Where k is the DC-gain and τ is the time-constant. The system has a pole in $s = -\frac{1}{\tau}$ i.e., the smaller time-constant, the faster system response.

2.1.2 Second Order Systems

The transfer function of a second-order system is given by:

$$H(s) = \frac{k\omega_n^2}{s^2 + 2\zeta\omega_n s + \omega_n^2}$$

Where $\omega_n > 0$ is the natural frequency and $\zeta > 0$ is the damping ratio and k is the gain.

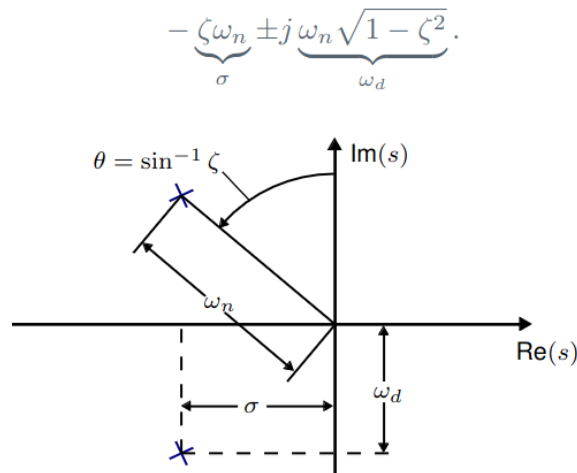
The system has two poles, which are $s \in \mathbb{C}$ where:

$$s^2 + 2\zeta\omega_n s + \omega_n^2 = 0$$

The values of s is given by:

$$s = -\zeta\omega_n \pm j\omega_n\sqrt{1-\zeta^2}$$

When $\zeta = 1$ the system is critically damped and $H(s)$ has a double pole in $s = -\zeta\omega_n$, when $0 < \zeta < 1$ the system is underdamped and has complex poles. When $\zeta > 1$ the system is overdamped and has real and distinct poles.



Impulse response of a underdamped second-order system:

$$h(t) = k \frac{\omega_n}{\sqrt{1 - \zeta^2}} e^{-\sigma t} \sin(\omega_d t) 1(t)$$

The step response of a underdamped second-order system:

$$y(t) = k(1 - e^{-\sigma t}(\cos(\omega_d t) + \frac{\sigma}{\omega_d} \sin(\omega_d t)))$$

Impulse response of a critically damped second order system:

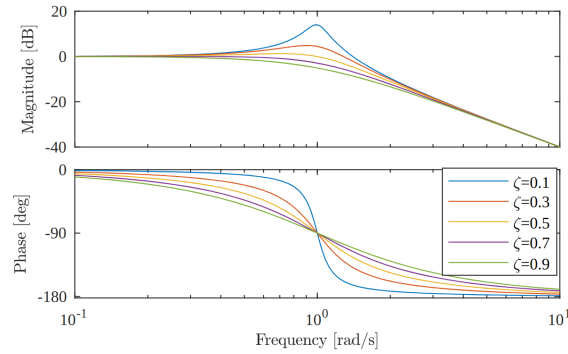
$$h(t) = k\omega_n^2 t e^{-\omega_n t}$$

Having a overdamped system with a damping ration that is greater than one leads to a slower impulse response.

The step response of a critically damped second order system:

$$y(t) = k(1 - e^{-\omega_n t} - \omega_n t e^{-\omega_n t})$$

Looking at the bode plot of a second-order system depends on the damping ratio. For every pole the magnitude of the bode plot decreases by 20dB/decade. And the phase of the bode plot decreases by 90 degrees for every pole.

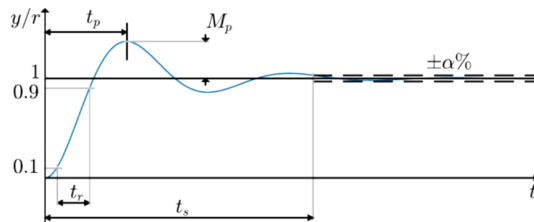


2.2 Performance specifications

Three different performance measurements of dynamical systems are:

1. The rise time t_r (stigetid)
2. The settling time t_s (indsvingningstid)
3. The overshoot M_p (oversving)

The rise time is the time it takes for the system to go from 10% to 90% of the final value. The settling time is the time it takes for the system to reach and stay within a certain % of the final value. The overshoot is the maximum percentage that the system goes over the final value.



The peak time is denoted t_p which is the time where the signal reaches its maximum value.

For a second-order system the rise time can be approximated as:

$$t_r = \frac{1.8}{\omega_n}$$

For a second-order system the settling time can be approximated as:

$$t_s = \frac{-\log(\alpha/100)}{\zeta\omega_n}$$

Log is ln in the equation.

The peak time can be found using:

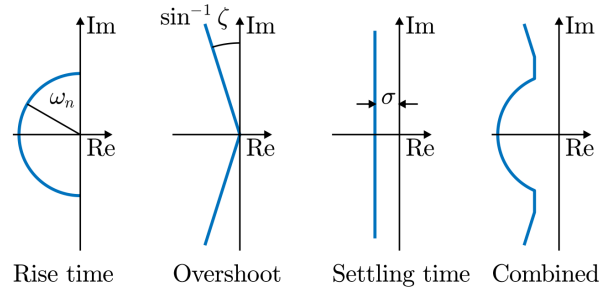
$$t_p = \frac{\pi}{\omega_d}$$

The overshoot is computed from the step response at the peak time. This gives an expression for the overshoot:

$$M_p = e^{-\frac{\zeta\pi}{\sqrt{1-\zeta^2}}}$$

for $0 < \zeta < 1$. The overshoot is given in percentage.

Overview for determining performance variables



To obtain a rise time shorter than t_r

$$\omega_n \geq \frac{1.8}{t_r}$$

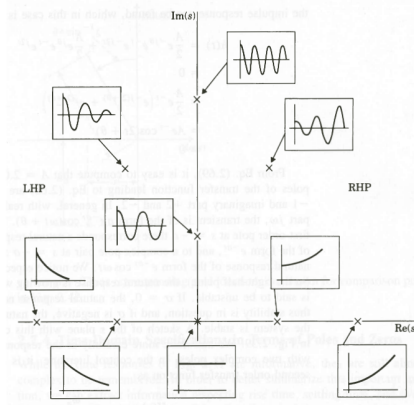
To obtain an overshoot that is smaller than M_p

$$\zeta \geq \sqrt{\frac{\left(\frac{\log(M_p)}{-\pi}\right)^2}{1 + \left(\frac{\log(M_p)}{\pi}\right)^2}}$$

To obtain an $\alpha\%$ -settling time shorter than t_s

$$\sigma \geq \frac{-\log(\alpha/100)}{t_s}$$

Overview of s plane



The waveform at a given point in the s -plane is given by e^{st} which means that

1. The oscillation frequency increases when $|\text{Im}(s)|$ increases.
2. The decay rate increases when $\text{Re}(s)$ decreases.

2.3 Poles and Zeros of Space-State Models

Using the state-space matrices:

$$G(s) = C(sI - A)^{-1}B + D$$

Poles can be found using the following equation:

When $G(s) \rightarrow \infty$ the system has a pole at $s \rightarrow p$.

$$\det(pI - A) = 0$$

Here p is the eigenvalue of A .

Zeros can be found using the following equation:

When $G(s) = 0$ the system has a zero at $s = z$.

$$\begin{vmatrix} A - zI & B \\ C & D \end{vmatrix} = 0$$

Another definition of z being a zero for $G(s)$ is when the matrix does not have full column rank.

2.4 Stability

The stability of the dynamical system can be determined from the eigenvalues of A in the time domain. This is the equivalent of poles in the frequency domain.

$$\dot{x} = Ax + Bu$$

When the eigenvalues of A have a negative real part, the system is stable.

Definition A linear discrete-time system

$$x_{k+1} = \Phi x_k$$

The state of the next time step is given by ϕ multiplied by x_k .

Where $x_k \in \mathbb{R}^{n \times n}$ is asymptotically (asymptotisk) stable if

$$\lim_{k \rightarrow \infty} x_k = 0$$

for any $x_0 \in \mathbb{R}^n$.

A linear discrete-time system is stable if all the eigenvalues are inside the unit circle.

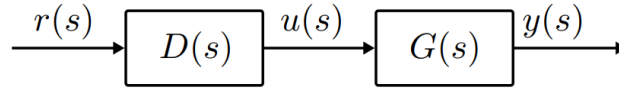
The closer the poles are to origin the more dominating the pole is in the system.

2.5 Examples

3 Introduction to Control

3.1 Open Loop Control

(åbensløjfe regulering)



Steady-State Value of Time Function:

Suppose that $Y(s)$ is the Laplace transform of $y(t)$. Then the final value of $y(t)$ is either:

- Unbounded. If $Y(s)$ has any poles in the open right half-plane (unstable)
- Undefined. If $Y(s)$ has a pole pair on the imaginary axis.
- Constant. If all poles of $Y(s)$ are in the open left half-plane, except for one at $s=0$

The Final Value Theorem (slutværdi-sætningen)

If all poles of $sY(s)$ are in the open left half-plane, then:

$$\lim_{t \rightarrow \infty} y(t) = \lim_{s \rightarrow 0} sY(s)$$

The Final Value Theorem determines the constant value that the impulse response of a stable system converges to. The theorem can also be used to determine the DC gain of a system, i.e., the output when a step input is applied to the system. When finding the DC gain of a system $G(s)$ then recall that the impulse response of the transfer function $1/s$ is a constant. Thus the DC gain is given by:

$$\lim_{s \rightarrow 0} sG(s) \frac{1}{s} = \lim_{s \rightarrow 0} G(s)$$

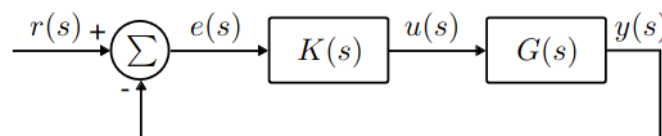
The reason as to $1/s$ is used is because you go from impulse response to step response by integrating the impulse response.

If the system has a disturbance an open loop control system will not be able to compensate for this.

3.2 Feedback Control

(tilbagekobling). Tilbagekobling anvendes for at eliminere forstyrrelsesundertrykkelse.

The connection of a controller $K(s)$ and a system (also called plant) $G(s)$ is called a closed-loop system.



$e(s)$ is the error signal, and is the difference between the reference signal $r(t)$ and the output signal $y(t)$. The error signal is used to adjust the input signal $u(t)$ to the system. The controller $K(s)$ is used to adjust the input signal.

The closed-loop transfer function is given by:

$$\frac{y(s)}{r(s)} = \frac{G(s)K(s)}{1 + G(s)K(s)}$$

$K(s)$ can be designed to meet certain requirements, such as stability, disturbance rejection, and sensitivity analysis.

Stability

The closed-loop system is stable if all the closed-loop poles are in the open left half-plane.

The **loop gain (sløjfe-forstærkning)** is defined as: $L(s) = G(s)K(s)$. The closed-loop poles are given by:

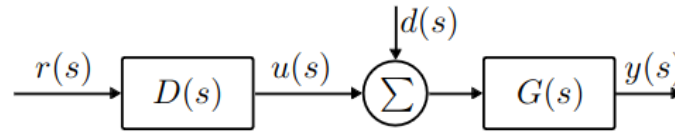
$$1 + L(s) = 0$$

The closed loop poles satisfy:

$$1 + L(s) = 0 \Rightarrow L(s) = -1$$

Disturbance Rejection

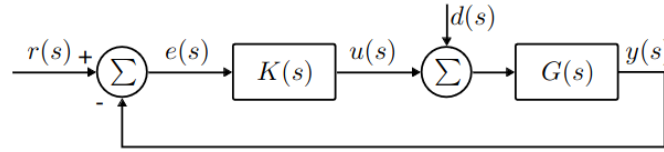
Open loop control system:



The output is given by:

$$y_m(s) = G(s)d(s) + G(s)D(s)r(s)$$

Closed loop control system:



The output is given by (superposition can be applied for deriving the expression):

$$y_m(s) = \frac{G(s)}{1 + G(s)K(s)}d(s) + \frac{G(s)K(s)}{1 + G(s)K(s)}r(s)$$

The larger the amplification of the feedback loop, the smaller the effect of the disturbance on the output.

Sensitivity Analysis

The sensitivity S of the open-loop system is given by the ratio of $\delta T_{ol}/T_{ol}$ to $\delta G_0/G_0$. For the open loop system $S=1$. δ is the relative change in the transfer function. This means that if the DC gain of the system changes by 10% then the output will change by 10%.

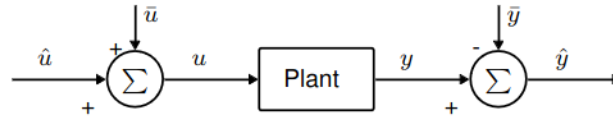
For the closed loop the sensitivity is given by:

$$S = \frac{1}{1 + G_0 K_0}$$

This means that a large amplification leads to a low sensitivity.

Linearization

The transfer function from \hat{u} to \hat{y} of the linearized model is linear.



To transform the linearized model into state space form, it is seen that

$$\hat{z}^{(n)} \approx \frac{\partial f}{\partial z} \bigg|_p \hat{z} + \dots + \frac{\partial f}{\partial z^{(n-1)}} \bigg|_p \hat{z}^{(n-1)} + \frac{\partial f}{\partial u} \bigg|_p \hat{u}$$

is on the standard form

$$\dot{z}^{(n)} - a_{n-1}z^{(n-1)} - \dots - a_1\dot{z} - a_0z = b_1u_1(t) + \dots + b_mu_m(t)$$

with

$$a_i = \frac{\partial f}{\partial z^{(i)}} \bigg|_p \quad \text{and} \quad b_i = \frac{\partial f}{\partial u_i} \bigg|_p$$

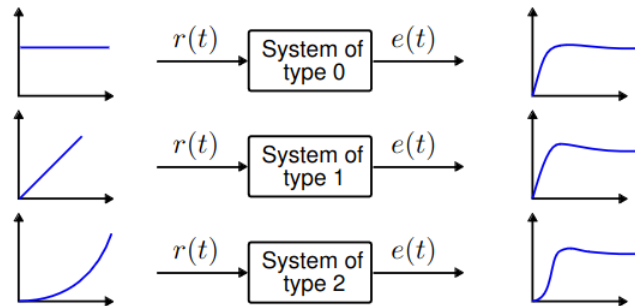
Then the following procedure from Lecture 1 can be followed, by defining the state variables x_i according to $x_1 = \hat{z}$, $x_2 = \hat{z}^{(1)}$, \dots , $x_n = \hat{z}^{(n-1)}$.

3.3 Steady State Tracking

Referencefølge

Definition of system type

Stable systems can be classified according to its system type, defined to be the degree of the polynomial for which the steady-state error in a nonzero finite constant.



If you want zero steady state error to a constant reference signal, then the system must be of type 1. However, if you apply a ramp signal as a reference signal to a type 0 system, then the steady state error will be infinite. Thus, you will have a static error if the system type matches the input type as shown in the previous figure.

An expression for the tracing error is computed to determine the steady-state error. The error is given by:

$$e(s) = \frac{1}{1 + L(s)} r(s)$$

Where $r(s)$ is the reference signal and $L(s)$ is the loop gain.

Final value theorem is used to determine the steady-state error when the reference signal is a polynomial $r(t) = t^k 1(t)$. Where k is the input type.

$$e_{ss} = \lim_{t \rightarrow \infty} e(t) = \lim_{s \rightarrow 0} e(s)s$$

$$\lim_{s \rightarrow 0} e(s)s = \lim_{s \rightarrow 0} s \frac{1}{1 + L(s)} r(s)$$

$$\lim_{s \rightarrow 0} s \frac{1}{1 + L(s)} \frac{1}{s^{k+1}}$$

The expression for the steady state error is simplified by rewriting the loop gain as:

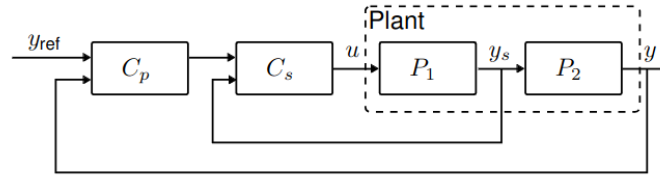
$$L(s) = \frac{L_0(s)}{s^n}$$

where n is the number of poles at the origin of L(s). This means that the amount of poles at s=0 of the systems loop gain L(s) determines the system type.

Type	Input		
	Step (Position)	Ramp (Velocity)	Parabola (Acceleration)
Type 0	$\frac{1}{1+K_p}$	∞	∞
Type 1	0	$\frac{1}{K_v}$	∞
Type 2	0	0	$\frac{1}{K_a}$

3.4 Cascade Control

A cascade control uses the output from one controller as the input to another controller.



The nested control loops are called the inner loop (secondary) and the outer loop (primary). A fundamental reason for applying cascade control is to obtain better disturbance rejection and lower sensitivity to parameter variations.

Disturbance rejection

Single loop control with two plants:

$$y(s) = P_2(s)P_1(s)d(s) + P_2(s)P_1(s)u'(s)$$

Cascade control with two plants and two control loops:

$$y(s) = \frac{P_2(s)P_1(s)}{1 + P_1(s)C_s(s)}d(s) + \frac{P_2(s)P_1(s)C_s(s)}{1 + P_1(s)C_s(s)}u'(s)$$

where $C_s(s)$ is the controller for the secondary loop.

The inner loop should be faster than the outer loop. A rule of thumb is that the average residence times should have a ratio of at least five. Essential disturbances should act in the inner loop.

3.5 DC Motor Dynamics

3.6 Examples

4 Design of PID Controllers

4.1 P-controller

The control law for the proportional controller is given by

$$u(t) = K_p e(t)$$

where K_p is the proportional gain and $e(t)$ is the error signal.

The controller applies a control signal to the system, which depends linearly on the error signal.

By increasing the gain, the steady state error gets smaller, but the overshoot gets larger. Moreover, the system gets a faster response.

To eliminate the steady state error, feedforward can be added to the P-controller.

$$u(t) = K_p e(t) + u_{ff}$$

Where the term u_{ff} is also called reset. We choose the feedforward according to the DC gain of the system, i.e.

$$u(t) = K_p e(t) + \frac{1}{G(0)} r(t)$$

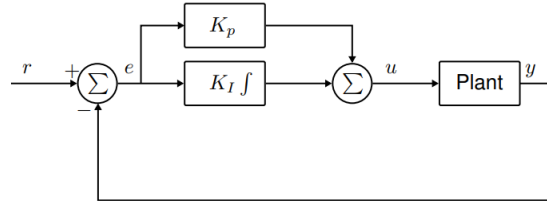
The feedforward does not affect the poles of the closed loop system. Remember that feedforward does not eliminate noise disturbances. In addition to this if the model is not perfect, the feedforward control will not work.

4.2 PI-controller

The control law of a proportional-integral feedback controller is given by:

$$u = K_p e + K_I \int_{t_0}^t e(\tau) d\tau$$

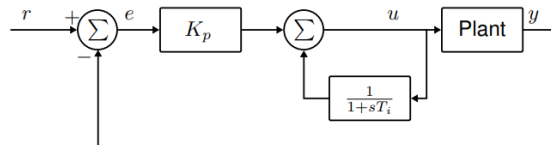
The controller adds a term, which is proportional to the integral of the error e to the P-control term.



The purpose of adding the integral term is to eliminate the steady state error of the system without the need for feed forward, i.e., integral control is less sensitive to modelling errors. Feed forward cannot eliminate modelling errors, but integral control can.

Comparing the PI-controller to the P-controller, the PI-controller has a better response to disturbances, that are unknown or cannot be measured. This is because the feedforward control can only eliminate the steady state error if the model and disturbances are known.

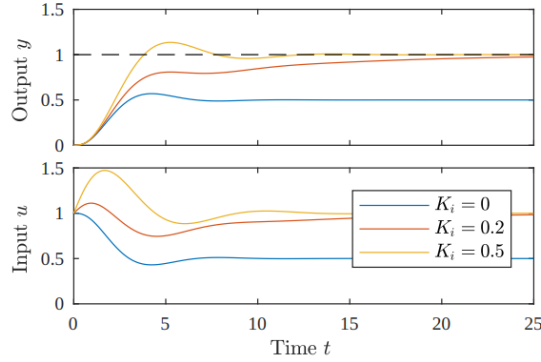
Implementation of a PI-controller is called automatic reset, and is shown in the following figure.



On the figure, T_i is the integration time (integraltid). The transfer function from e to u is given by:

$$T_{ue} = K_p \frac{1 + sT_i}{sT_i} = K_p + \frac{K_p}{sT_i}$$

On the figure below the step responses are shown for $K_i = 0, 0.2, 0.5$ and $K_p = 1$



From the unit step response it can be seen that the integral action removes the steady state error. A large K_i will give a fast response, but also a large overshoot. A small K_i will give a slow response, but also a small overshoot.

4.3 PD-controller

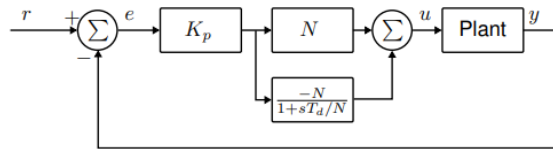
The derivative term is used to decrease oscillations and overshoots in the system. But having a large d-term can also lead to a slow settling time.

The derivative term provides an anticipatory action to the control, by doing feedback based on the trend of the error, i.e.

$$u(t) = K_p e(t) + K_d \frac{de(t)}{dt} = K_p \left(e + T_d \frac{de(t)}{dt} \right)$$

Where k_d is the derivative gain and T_d is the derivative time constant and e_p is a prediction of the error (T_d forwards in time). The parenthesis in the equation on the right is a prediction of the error called e_p .

PD-implementation



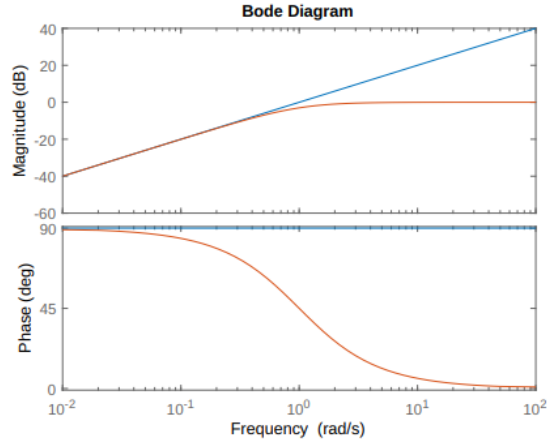
Noise is an issue for controllers that include a derivative term. Therefore, they can be implemented in a low-pass filtered version.

The transfer function of the controller is

$$T_{ue}(s) = K_p \left(N - \frac{N}{1 + sT_d/N} \right) = K_p \frac{sT_d}{1 + sT_d/N}$$

Where N is a filter constant (typical values of N are 2 to 20), and T_d is the derivative time constant.

The bode plot of an ideal PD-controller and a filtered PD controller are similar for low frequencies. Using a low-pass filter, we can at a certain frequency, reduce the gain of the derivative term, and thereby reduce the noise amplification.

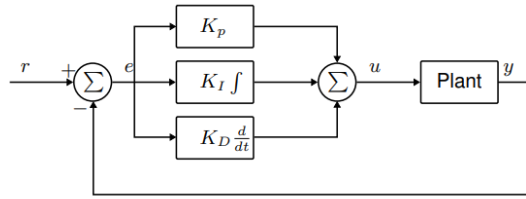


4.4 PID-controller

The control law of a PID feedback controller is

$$u(t) = K_p e(t) + K_i \int_{t_0}^t e(\tau) d\tau + K_d \frac{de(t)}{dt}$$

A block diagram of the controller is given below:



It can also be written as:

$$K(s) = K_p \left(1 + \frac{1}{sT_i} + sT_d \right)$$

Alternatively, the PID-controller with filter on the D-term is

$$K(s) = K_p \left(1 + \frac{1}{T_i s} + \frac{sT_d}{1 + sT_d/N} \right)$$

4.5 Tuning a PID Controller

4.5.1 Pole placement

If a model of the system is available, then it is possible to compute an expression for the characteristic polynomial of the closed-loop system. Based on this polynomial, it may be possible to place the poles at desired locations.

4.5.2 Ziegler-Nichols method

Sometimes a model of the plant is not available, then the controller should be tuned by only studying the input-output behaviour of the system. Ziegler and Nichols has proposed two methods for tuning PID controllers without explicit use of a plant model.

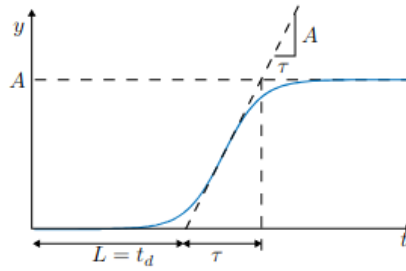
- Ziegler-Nichols tuning based on step response
- Ziegler-Nichols tuning - The ultimate sensitivity method

Tuning based on step response

The considered system is assumed to have the transfer function

$$\frac{y(s)}{u(s)} = \frac{A}{\tau s + 1} e^{-st_d}$$

This is a first-order system with a time delay of t_d .



The parameters for the PID controller can be found using the following table:

Controller type	Gains
P	$K_p = \frac{1}{RL}$
PI	$K_p = \frac{0.9}{RL}$ $T_i = \frac{L}{0.3}$
PID	$K_p = \frac{1.2}{RL}$ $T_i = 2L$ $T_d = 0.5L$

where $R = A/\tau$ is the amplitude of the oscillation, and $L = t_d$. Remember that the table above is only an initial guess, and the controller should be tuned further.

T_i is the time integration constant. The constant describes how fast the integral term is accumulated. A large T_i will give a slow response, but a small overshoot. A small T_i will give a fast response, but a large overshoot. T_d is the derivative time constant. The constant describes the period over which the derivative term is calculated. A large T_d will enhance the controller's responsiveness to changes in error, but at the risk of amplifying noise. T_d is used to predict and counteract future errors. Whereas the T_i term is used to eliminate steady state errors.

The ultimate sensitivity method

Start using a P-controller, and increase the gain until the system starts to oscillate. The value of K_p when the output oscillates with a constant amplitude is called the ultimate gain K_u . The period of the oscillation is called the ultimate period P_u .

The parameters for the PID controller can be found using the following table:

Controller type	Gains
P	$K_p = 0.5K_u$
PI	$K_p = 0.45K_u$ $T_i = \frac{P_u}{1.2}$
PID	$K_p = 0.6K_u$ $T_i = 0.5K_u$ $T_d = \frac{1}{8}P_u$

Values of the table should also only be used as an initial guess for the controller gains. The controller most often needs manual tuning as well.

4.6 Examples

5 Root Locus

Rudkurve metoden. Grafisk metode til desgin af regulatorer.

Rule 1

Consider the characteristic equation:

$$1 + KG(s) = 1 + K \frac{Q(s)}{P(s)} = 0$$

This can be written as:

$$P(s) + KQ(s) = 0$$

This is a polynoial of degree $N = \max(m, n)$ where m is the number of poles and n is the number of zeros. Lemma. A univariate polynomial of degree d has d roots in \mathbb{C} .

There are N lines (loci) where $N = \max(m, n)$. Where m is the number of poles and n is the number of zeros.

Rule 2

Similar to previously, the characteristic equation is rewritten as:

$$P(s) + KQ(s) = 0$$

Let $K = 0$, then we observe that the roots of the characteristic equation are the poles of the open-loop system.

Let $K \rightarrow \infty$, then we observe that the roots of the characteristic equation are the zeros of the open-loop system.

$$\frac{P(s)}{K} + Q(s) = 0$$

Defintion: As K increases from 0 to ∞ , the closed loop roots move from the open loop poles of $G(s)$ to the open-loop zeros of $G(s)$.

Rule 3

When roots are complex they occur in conjugate pairs.

Rule 4

We study the rewritten characteristic equation:

$$\frac{Q(s)}{P(s)} = -\frac{1}{K}$$

And see that the phase of $\frac{Q(s)}{P(s)}$ is 180° to satisfy the equation.

The transfer function can be written as:

$$T(s) = \frac{Q(s)}{P(s)} = \frac{(s - z_1)(s - z_2) \cdots (s - z_m)}{(s - p_1)(s - p_2) \cdots (s - p_n)}$$

where z_i are the zeros and p_i are the poles. Let $z \in \mathbb{C}$ then $zz^* = |z|^2$; hence, complex pole pairs and pairs of complex conjugated zeros do not affect the phase of $T(s)$ for $s \in \mathbb{R}$.

The phase of $(s - z_m)$ when $s, z_m \in \mathbb{R}$ is

$$\angle(s - z_m) = \begin{cases} 180^\circ & \text{if } s > z_m \\ 0^\circ & \text{otherwise} \end{cases} \quad (1)$$

The portion of the real axis to the left of an odd number of open loop poles and zeros are part of the loci.

Rule 5

Lines leave and enter the real axis at 90° angles.

Rule 6

For very large values of s the equation:

$$1 + K \frac{s^m + b_1 s^{m-1} + \dots + b_m}{s^n + a_1 s^{n-1} + \dots + a_n} = 0$$

can be approximated to:

$$1 + K \frac{1}{(s - \alpha)^{n-m}}$$

The phase of this expression should be 180° for this expression to hold. This implies that:

$$(n - m)\phi_l = 180^\circ + 360^\circ(l - 1)$$

where ϕ_l is the phase of the expression.

$$\phi_l = \frac{180^\circ + 360^\circ(l - 1)}{n - m}$$

let $m < n - 1$ then:

$$-\sum r_i = -\sum p_i$$

where r_i are the closed-loop poles and p_i are the open-loop poles.

For s going to ∞ , it is known that m closed-loop poles go towards the open-loop zeros, and $n - m$ closed-loop poles go towards α , i.e.

$$-\sum r_i = -(n - m)\alpha - \sum z_i = -\sum p_i$$

where z_i are the open-loop zeros.

Thus,

$$\alpha = \frac{\sum z_i - \sum p_i}{n - m}$$

5.1 Examples

Departure angle

q = multiplicity of the pole. This means that the pole is repeated q times at the same location.

6 The Nyquist Stability Criterion

6.1 Frequency response

The frequency response of a system is the steady-state response of a system to a sinusoidal input. The output of a time-invariant system will have the same frequency as the sinusoidal input, but possibly with a different amplitude and phase.

The output of a time-invariant system is given by:

$$y(t) = \int_{-\infty}^{\infty} h(\tau)u(t - \tau)d\tau$$

Where y is the output, u is the input, and h is the impulse response of the system.

To obtain the frequency response, only sinusoidal inputs are considered.

The frequency response of $H(s)$ is given by the magnitude and phase of $H(j\omega)$

$$M = |H(j\omega)|$$

$$\phi = \angle H(j\omega)$$

The M is the amplitude ratio and ϕ is the phase shift.

The system response to the input $u(t) = \cos(\omega t)$ is given by:

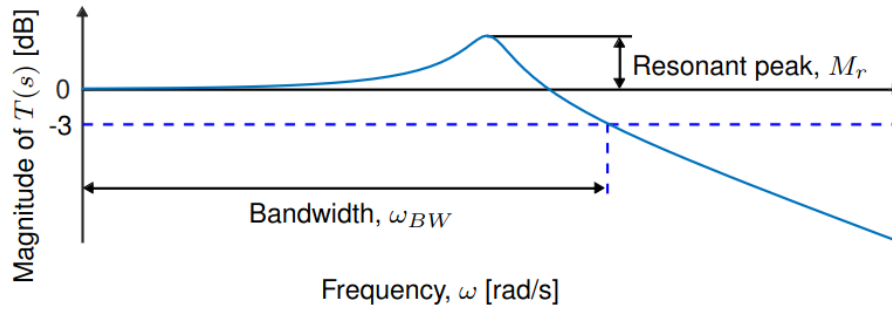
$$y(t) = \frac{A}{2}M(\omega)\cos(\omega t + \phi(\omega))$$

This explains why the output will have the same frequency as the input, but possibly with a different amplitude and phase.

The bandwidth of a closed-loop system $T(s)$ is defined to be the maximum frequency at which the output y of a system will track a sinusoidal input r in a satisfactory manner. Output attenuated to $1/\sqrt{2}$ of the input amplitude. Formally the bandwidth ω_{BW} of $T(s)$ is the maximal frequency such that:

$$|T(j\omega)| \geq 1/\sqrt{2}$$

The maximal value of the frequency response is called the resonant peak,



6.2 Bode Plots

Pole decreases the magnitude of the frequency response by 20dB/decade. Zero increases the magnitude of the frequency response by 20dB/decade.

Bode Form of transfer function

In the following, we consider the open-loop transfer function

$$KG(s) = K \frac{(s - z_1)(s - z_2) \cdots}{(s - p_1)(s - p_2) \cdots}$$

For convenience, the transfer function is evaluated at $s = j\omega$ and rewritten into Bode form as follows:

$$KG(j\omega) = K_0 \frac{(j\omega\tau_1 + 1)(j\omega\tau_2 + 1) \cdots}{(j\omega\tau_a + 1)(j\omega\tau_b + 1) \cdots}$$

Classes of terms in transfer function

Class 1: $K_0(j\omega)^n$ for $n \in \mathbb{Z}$

If $n > 0$ then the term is a zero at the origin. If $n < 0$ then the term is a pole at the origin. It can be more than one depending on the value of n .

Example: For the term $K_0(j\omega)^n$ we have:

$$\log K_0 |j\omega| = \log K_0 + n \log \omega$$

This means that the magnitude plot is a straight line with slope $n \cdot (20\text{dB/dec})$. The phase of $(j\omega)^n$ is constant and given by: $\phi = n \cdot 90^\circ$

If $n = -1$, this means that there is a pole at the origin. This therefore leads to a slope of -20dB/dec and a phase shift of -90°

Class 2: $(j\omega\tau + 1)^{\pm 1}$

real pole or zero.

For $\omega\tau \ll 1, j\omega\tau \approx 1$

For $\omega\tau \gg 1, j\omega\tau + 1 \approx j\omega\tau$

In addition, for $\omega = 1/\tau$, the gain is $\sqrt{2}$ - an increase of 3dB compared to the DC gain. The point $\omega = 1/\tau$ is called the break point. At the break point the angle is 45° . Thus, the phase changes from a decade before the break point to a decade after the break point.

Class 3: $((j\omega/\omega_n)^2 + 2\zeta(j\omega/\omega_n) + 1)^{\pm 1}$

Complex conjugate poles or zeros.

For $\omega \ll \omega_n$, the amplitude is approximately 1. In addition, at the break point $\omega = \omega_n$, the magnitude is $|G(j\omega)| = 1/(2\zeta)$ and the phase is $\pm 90^\circ$

Summary of Bode Plot Rules

1. Rewrite the considered transfer function to bode form
2. Determine the value of the $K_0(j\omega)^n$ term. Plot the low frequency magnitude asymptote through the point K_0 at $\omega = 1$ and with slope of $n \cdot 20\text{dB/dec}$
3. Complete the composite magnitude asymptotes by extending the low-frequency asymptote until the first frequency break point. Then change the slope according to the behavior at the break point, and continue the procedure for the remaining break points.
4. Sketch the approximate magnitude curve by increasing the asymptote value by a factor $\sqrt{2}$ at first-order numerator break and decreasing it by a factor $1/\sqrt{2}$ at denominator break.

5. Plot the low-frequency asymptote of the phase $\phi = n \cdot 90^\circ$
6. Change the phase at the phase points, and correct the phase according to the slope at the phase point.

6.3 Nyquist Stability Criterion

Nyquist can be used for open-loop unstable systems, whereas you can't use bode plots. But you must know the open loop poles in order to determine the number of closed-loop zeros in the right half plane.

A contour map of a complex function will encircle the origin $Z-P$ times where Z is the number of zeros and P is the number of poles of the function inside the contour.

To verify the stability of a system, one needs to determine the number of closed-loop poles in the right half plane. Thus, the Nyquist plot is a map of a contour that encircles the entire right-half plane.

The number of closed-loop poles in the right half-plane equals the number of right half-plane zeros of the characteristic equation. The characteristic equation is given by:

$$1 + KG(s) = 0$$

Let N denote the number of clockwise encirclements of -1. Then the number of zeros in the right half plane Z (closed-loop poles) minus the number of open-loop poles in the right half plane P .

$$N = Z - P$$

Steps

1. Nyquist plot of $KG(j\omega)$
2. Determine the number of encirclements of -1 and note direction
3. Figure out how many more poles or zeros are in the right half plane.

6.4 Examples

7 Dynamic Compensators and Stability Margins

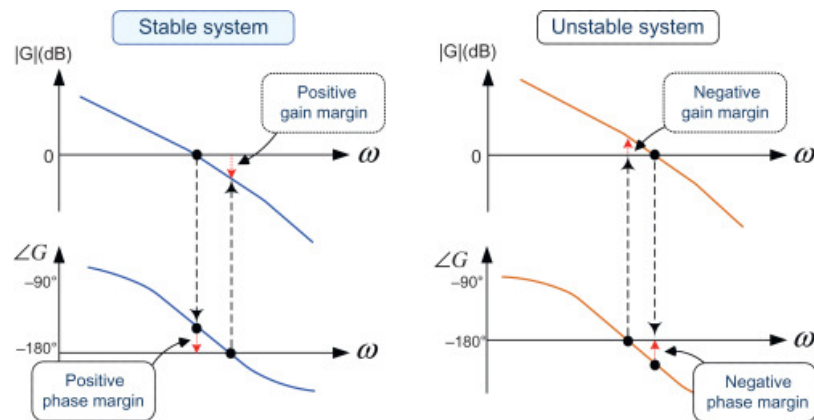
7.1 Stability Margins

Gain margin: factor by which the gain can be raised before a system becomes unstable. An uncertainty that may affect the gain of the system could be inertia.

Phase margin: The amount by which the phase can be increased before the system becomes unstable. (exceeds -180°) An uncertainty that may affect the phase of the system could be communication delays between a controller and a sensor.

For a system to be robust the gain margin should be 3 to 6dB and the phase margin should be 30° to 45° .

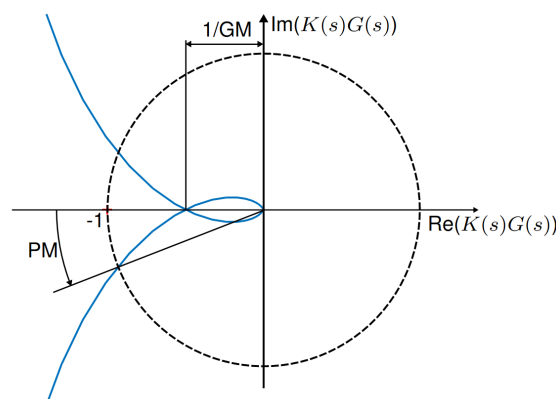
Remember when finding the phase and gain margin one should use the loop gain $L(s)$. The phase margin can be found on a bode plot by looking at the phase at the frequency where the gain is 0dB. The phase margin is then 180° minus the phase at that frequency. The gain margin is the gain at the frequency where the phase is -180° .



Neutral stability for loop gain $L(s)$ is when $L(s) = -1$. This is the same as a gain of 1 and a phase of -180° .

Nyquist

When we encircle -1 then you will have a pole in the right half plane. From the plot below it can be seen that the plot can be rotated the degrees that the phase margin is. The gain margin can be interpreted as seen on the figure below.



The advantage of using a nyquist plot compared to a bode plot is that you can see the phase and gain margin at the same time.

7.2 Dynamic compensation

Lead compensator is equivalent to a d-part for a PID controller. A lead compensator has a positive phase. A lead compensator decreases the rise time and overshoot.

Lag compensator is equivalent to an i-part for a PID controller. The lag controller improves the steady state tracking.

Both compensators are given by the transfer function:

$$D(s) = K \frac{s + z}{s + p}$$

Where z and p are the zero and pole of the transfer function respectively.

- If $z < p$, then $D(s)$ is called a lead compensation
- If $z > p$, then $D(s)$ is called a lag compensation

7.2.1 Lead compensation

Used for dynamic properties such as rise time, overshoot and settling time. It has relation to the phase margin, where we can lift the phase margin by adding a lead compensator.

A lead compensation is given by:

$$D(s) = \frac{Ts + 1}{\alpha Ts + 1}$$

and $1/\alpha$ is called the lead ratio.

We have the following properties:

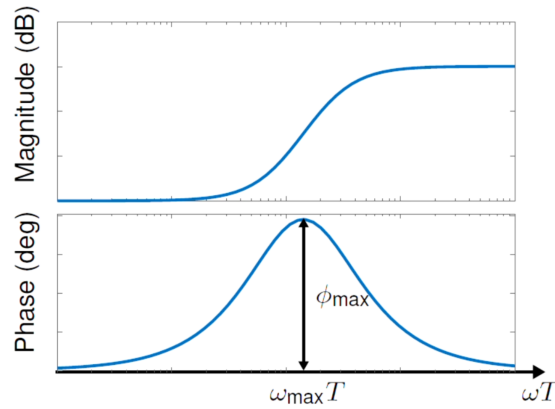
$$\omega_{max} = \frac{1}{T\sqrt{\alpha}} = \sqrt{|z||p|}$$

,

$$T = \frac{1}{\omega_{max}\sqrt{\alpha}}$$

$$\sin(\phi_{max}) = \frac{1 - \alpha}{1 + \alpha}$$

ϕ_{max} is the maximum phase lead. Which means that it is the maximum phase that the lead compensator can provide. And ω_{max} is the frequency at which the phase lead is maximum. Allow for extra margin about 10° , and one lead compensation should contribute a maximum of 60° to the phase.



7.2.2 Lag compensation

A lag compensator can be used for minimizing the steady state error without affecting the other dynamics. Only looking at stationary error. It doesn't eliminate the error but reduces it.

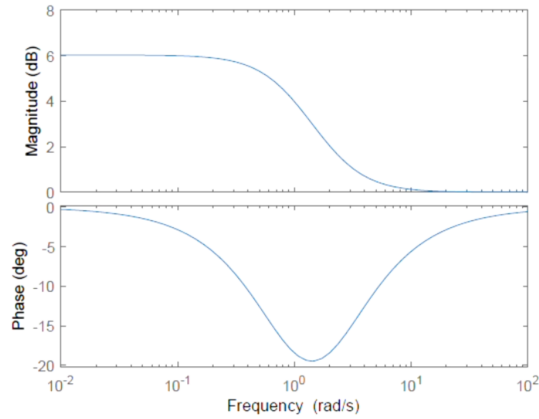
For a lag compensator the pole comes before the zero.

A lag compensator can be written as:

$$D(s) = K_0 \frac{\alpha Ts + 1}{Ts + 1}$$

where K_0 is the gain of the compensator, $z > p \in \mathbb{R}$ and $\alpha > 1$.

The gain decreases and the phase gets negative.



7.3 Examples

Lead compensation

Phase margin is found at 1.68 rad/s to be 7° .

To allow a phase margin of 25° plus 10 for extra margin the needed phase lead is:

$$\phi_{max} = 25 + 10 - 7 = 28^\circ$$

The value of α is found by:

$$\alpha = \frac{1 - \sin(28)}{1 + \sin(28)} = 0.3610$$

Pick ω_{max} to be at the crossover frequency, i.e., $\omega_{max} = 1.68$ rad/s. Then the value of T is:

$$T = \frac{1}{\omega_{max} \sqrt{\alpha}} = 0.9906$$

Test the new system and verify. This shows that the phase margin is too small. Thus, you should reiterate by moving the pole and zero.

8 Implementation

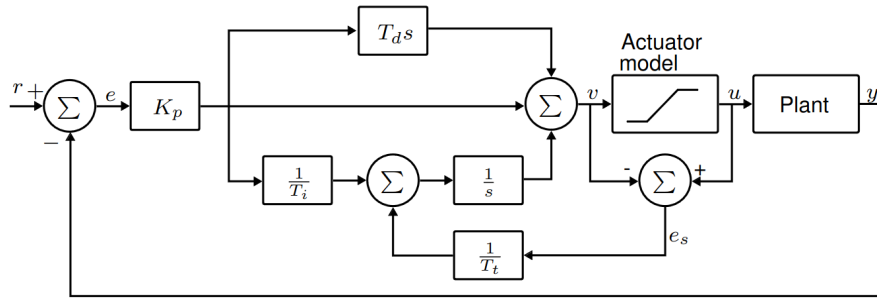
8.1 Anti-windup

Real electromechanical systems has saturations on their physical variables. In particular, the output of any actuator is limited from above and below. (Clipping). If you stay between the limits, the system will behave as a linear system. An example can be that a motor specified to use 24V, will not be able to handle 30V.

What happens to a PID controller when the system is saturated? The integrator will keep integrating the error, which will cause a large overshoot when the system is no longer saturated. This is called an integrator windup. To avoid this, anti-windup schemes are introduced.

Back-calculation

An anti-windup scheme that recomputes the integral term when the system input is saturated. The value of the integrator output is not changed instantaneously, but it is changed based on the tracking time constant.



The input to the integrator is given by

$$\frac{1}{T_t}e_s + \frac{K_p}{T_i}e$$

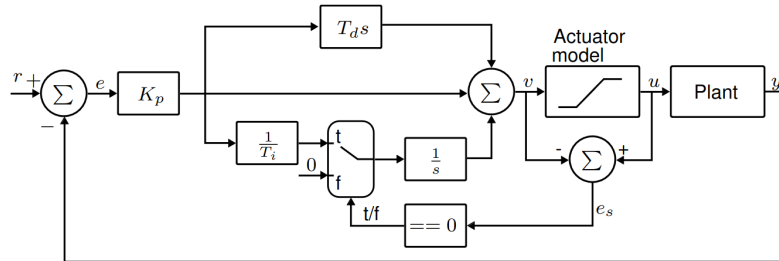
Where e_s is zero when the system is not saturated. In steady state, the output of the integrator is constant: hence, its input must be zero, i.e.

$$e_s = -\frac{K_p T_t}{T_i}e$$

If T_t is chosen to be small, the integrator will respond quickly to changes in the system.

Conditional integration

Also known as clamping is a bit simpler, and just stops integrating when the system is in saturation.



Setpoint weighting

To avoid large control signals when changing the reference rapidly, setpoint weights can be introduced. These modify the PID controller to be:

$$u(t) = K_p(\beta r(t) - y(t)) + K_i \int_0^t (r(\tau) - y(\tau)) d\tau + K_d(\gamma \frac{dr}{dt} - \frac{dy}{dt})$$

where γ and β are setpoint weights. Typically $\gamma = 0$ and β takes values between 0 and 1.

Filtering

To avoid large control signal noise on the measured output y , one adds a filter on the derivative term. Often a first order filter is sufficient, but this is application dependent.

$$u_d = k_d s \frac{1}{1 + sT_f}$$

Sometimes it is favorable to filter the control signal directly, thus the controller gets the form

$$C(s) = K_p(1 + \frac{1}{sT_i} + sT_d) \frac{1}{1 + sT_f + (sT_f)^2/2}$$

8.2 Digitalization

Most control systems are sample data systems, i.e., they consist of both discrete and continuous signals. The sampling frequency used for the discrete controller should be above 20 times the closed-loop bandwidth.

8.2.1 Emulation

Design continuous controller $K(s)$ and approximate it with $K(z)$ obtained via e.g. Tustin's method. Tustin's method also called the trapezoidal rule means to replace the variable s with

$$\frac{2}{T} \frac{z - 1}{z + 1}$$

The control output of the discrete PID controller can be written using 3 terms.

$$u_p(kT + T) = k_p e(kT + T)$$

$$u_I(kT + T) = u_I(kT) + K_i \frac{T}{2} (e(kT + T) + e(kT))$$

$$u_D(kT + T) = k_D \frac{2}{T} (e(kT + T) - e(kT)) - u_D(kT)$$

$$u(kT + T) = u_p(kT + T) + u_I(kT + T) + u_D(kT + T)$$

To analyze the system, the I-term and D-term are z-transformed:

$$zu_I(z) = u_I(z) + k_i \frac{T}{2} (ze(z) + e(z))$$

$$zu_D(z) = k_D \frac{2}{T} (ze(z) - e(z)) - u_D(z)$$

This gives the following expression for the controller

$$u(z) = (k_P + k_I \frac{T}{2} \frac{z + 1}{z - 1} + k_D \frac{2}{T} \frac{z - 1}{z + 1}) e(z)$$

Compensation for sampling effects

Design procedure

A discrete controller can be designed by emulation for the system $G(s)$ according to the next procedure:

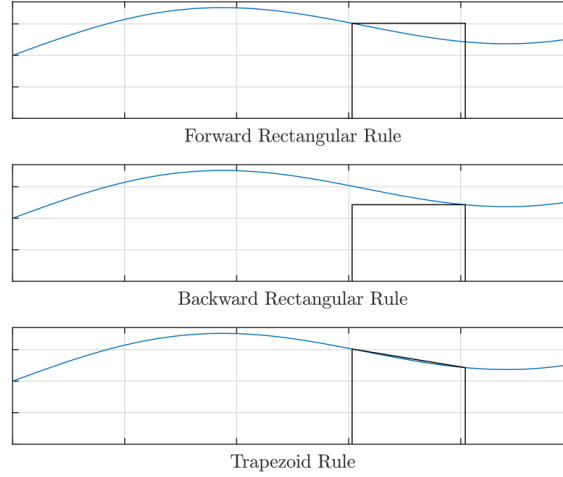
1. Design continuous compensation for the system $G_d(s)G(s)$, where $G_d(s)$ approximates a delay of $T/2$. G_d should be a first-order system with a time constant of $T/2$. Thus,

$$G_d(s) = \frac{1}{T/2s + 1}$$

2. Derive the discrete controller by applying Tustin's rule or the matched pole-zero method
3. Analyze the design by simulation or experimentally

8.2.2 Numerical Integration Methods

There are three methods to convert a continuous controller to a discrete controller:



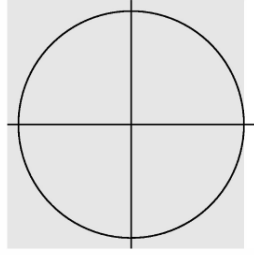
Get an approximation of the discrete transfer function by replacing s with:

Method:	Approximation:
Forward rule:	$s \leftarrow \frac{z-1}{T}$
Backward rule:	$s \leftarrow \frac{z-1}{Tz}$
Trapezoidal rule:	$s \leftarrow \frac{2}{T} \frac{z-1}{z+1}$

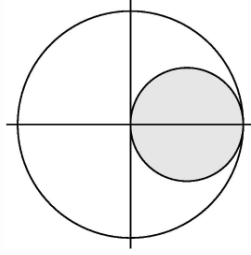
Get an approximation of the discrete transfer function by replacing z with:

Method:	Approximation:
Forward rectangular rule:	$z \leftarrow 1 + Ts$
Backward rectangular rule:	$z \leftarrow \frac{1}{1-Ts}$
Bilinear rule:	$z \leftarrow \frac{1+Ts/2}{1-Ts/2}$

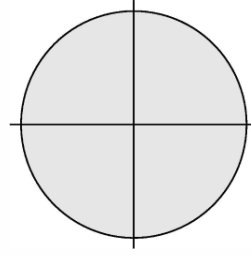
The left half-plane is mapped to different regions in the z -plane for the different methods.



Forward Rectangular Rule



Backward Rectangular Rule



Trapezoid Rule

This means that the discretization of a stable system using the forward rule can result in an unstable system.

8.2.3 Discrete Design

Design the discrete controller directly, without computing $K(s)$ first. Therefore it relies on a discretized plant model. The discrete transfer function of a system $G(s)$ and preceding zero-order hold is

$$G(z) = (1 - z^{-1})Z \left\{ \frac{G(s)}{s} \right\}$$

where Z denotes the Z-transform.

The system can subsequently be analyzed based on the closed-loop transfer function:

$$T_{cl}(z) = \frac{G(z)K(z)}{1 + G(z)K(z)}$$

Since the structure of the continuous and discrete closed-loop transfer functions are the same, the stability of the system can be studied via the characteristic equation:

$$1 + G(z)K(z) = 0$$

Consequently, the controller $K(z)$ can be designed via root locus or any of the other methods.

Design procedure

1. Transform the continuous-time plant into discrete time as follows:

$$G_d(z) = (1 - z^{-1})Z \left\{ \frac{G(s)}{s} \right\}$$

2. Design the feedback controller $K(z)$ using the same approaches as for a continuous-time.
3. Verify the design on the sampled data system.

8.3 Examples

Emulation example of PID controller

$$K(s) = 1.4 \frac{s+6}{s}$$

Use Tustin's method to emulate the controller in discrete time.

$$K(z) = K\left(\frac{2}{T} \frac{z-1}{z+1}\right) = 1.4 \frac{\frac{2}{T} \frac{z-1}{z+1} + 6}{\frac{2}{T} \frac{z-1}{z+1}} = 1.4 \frac{(1+3T)z - (3T-1)}{z-1}$$

In practice you implement the difference equation - not a discrete transfer function.

9 State Feedback Control

Tilstandsregulering (moderne regulering).

9.1 Controllability (styrbarhed)

Means if you can control your system to any desired state.

We cannot design a controller for a system that is not controllable in this course.

A continuous time system

$$\dot{x} = Ax(t) + Bu(t), x(0) = 0$$

is said to be controllable if and only if for any $\zeta \in \mathbb{R}^n$, there exists $u(t)$ such that for some $T > 0$, $x(T) = \zeta$

A discrete time system

$$x_{k+1} = \Phi x_k + \Gamma u_k, x_0 = 0$$

is said to be controllable if and only if for any $\zeta \in \mathbb{R}^n$, there exists (u_0, u_1, \dots) such that for some $N > 0$, $x_N = \zeta$

From iterating through the discrete time system, we get:

$$x_n = \Phi^{n-1}\Gamma u_0 + \dots + \Phi\Gamma u_{n-2} + \Gamma u_{n-1}$$

The controllability matrix (styrbarheds matricen):

$$x_n = \begin{bmatrix} \Gamma & \Phi\Gamma & \dots & \Phi^{n-1}\Gamma \end{bmatrix}$$

A system where $x \in \mathbb{R}^n, u \in \mathbb{R}^m$ is controllable if and only if the controllability matrix has full rank.

In continuous time you look at the matrix:

$$\text{Rank} \begin{bmatrix} B & AB & \dots & A^{n-1}B \end{bmatrix} = n$$

Having only 1 input signal, the system is controllable if:

$$\det \begin{Bmatrix} B & AB & \dots & A^{n-1}B \end{Bmatrix} \neq 0$$

9.2 Controllable Canonical Form

Any controllable system can be transformed into a controllable canonical form. The reason as to doing so is to simplify the design of the state feedback controller.

Considering $n = 3$ and introduce the following notation for the rows of T^{-1} :

$$T^{-1} = \begin{bmatrix} s_1 \\ s_2 \\ s_3 \end{bmatrix}$$

$$A_c = T^{-1}AT$$

$$B_c = T^{-1}B$$

9.3 State Feedback and Pole Assignment

Tilstands regulator.

Eigen values of $(A + BF)$ is the poles of the closed loop system. We can change F so $A + BF$ has any desired eigen values.

If the system is controllable and placed on controllable canonical form, it is then possible to place the poles of the closed loop system at any desired location using F .

$$F_c = \begin{bmatrix} a_{cl,1} - a_1 & \dots & a_{cl,n} - a_n \end{bmatrix}$$
$$F = F_c T^{-1}$$

9.4 Examples

10 Observer Design

Is used when you can't measure all states.

10.1 Observability

Estimation of state.

Tells when it is possible to estimate the states of a system from the output. An observer can only be made if the system is observable.

Observability matrix

$$O = \begin{bmatrix} C \\ CA \\ CA^2 \\ \vdots \\ CA^{n-1} \end{bmatrix}$$

Has to have full rank to be observable.

10.2 Full Order Observer

hat symbolizes the estimated value.

Estimation of all states. (tilstandsestimator)

$$\begin{aligned}\dot{\hat{x}} &= A\hat{x} + Bu + L(C\hat{x} - y) \\ \hat{y} &= C\hat{x}\end{aligned}$$

Error, $e = \hat{x} - x$:

$$\begin{aligned}\dot{e} &= \dot{\hat{x}} - \dot{x} = A\hat{x} + Bu + L(C\hat{x} - y) - (Ax + Bu) \\ &= A(\hat{x} - x) + L(C\hat{x} - Cx) \\ &= (A + LC)e\end{aligned}$$

Eigenvalues for $A+LC$ should be placed in the left half plane for the error to converge to zero. This has to be done by only changing L .

Theorem

A full order observer for the system with observer gain L is stable, if and only if the eigenvalues of the matrix $A + LC$ all have a negative real part. Moreover, such an L always exists, if (A, C) is observable.

Observable Canonical Form

Any observable single output system can be written in the form:

10.3 Observer Design

Designing an observer needs an observer gain matrix L . This can be found by

10.4 Observer Based Control

Combination of observer and state feedback. This gives $2n$ poles because of the extra eigenvalues. The state feedback poles should be placed 2 to 6 times further to the left than the observer poles. This is done to achieve a faster response.

Seperation principle:

You can design the observer and the controller independently and then combine them. An observer with observer gain L and feedback gain F results in $2n$ closed loop poles, coinciding with the eigenvalues of $A + BF$ and $A + LC$.

10.5 Examples

11 Integral Control and Optimal Control

11.1 Integral Control

11.2 Anti-windup

Must always be applied when having an integrator in the controller.

Designing Saturation Gain

M is a compensator gain. The compensator gain is only used when $F * \hat{x}$ is in saturation.

Dynamics of controller during saturation

$$\dot{\hat{x}} = (A + LC + MF)\hat{x}$$

Determining M can be recognized as an observer gain design problem:

$$\dot{\hat{x}} = (\tilde{A} + \tilde{L}\tilde{C})\hat{x}$$

With $\tilde{A} = A + LC$, $\tilde{L} = M$, $\tilde{C} = F$. From which the unknown $\tilde{L} = M$ can be chosen to assign any desired poles to the saturated controller.

11.3 Optimal Control

Linear quadratic regulator (LQR). Optimal refers to speed and precision.

Consider a linear control system of the form:

$$\begin{aligned}\dot{x} &= Ax + Bu, x(0) = x_0 \\ y &= Cx\end{aligned}$$

A control law for such a system is said to be optimal, if it minimizes the cost functional:

$$J = \int_0^\infty (x^T Q x + u^T R u) dt$$

Where $Q = Q^T$ is a positive semi-definite matrix (eigenvalues larger than or equal to 0) and $R = R^T$ is a positive definite matrices. Positive definite means that the eigenvalues must be larger than 0. The cost is optimal if the equation is close to zero.

Algebraic Riccati Equation is a second order matrix equation in an indeterminate $P = P^T \in \mathbb{R}^{n \times n}$:

$$A^T P + PA - PBR^{-1}B^T P + Q = 0$$

P is called a stabilizing solution to the ARE, if it satisfies the equation, and further satisfies that the eigenvalues of $A - BR^{-1}B^T P$ have negative real parts.

The optimal state feedback law is given by:

$$u = Fx$$

Where F is given by: $F = -R^{-1}B^T P$

Brysons rule

Used to determine the diagonal matrices Q and R. The rule states that the optimal control law is given by:

$$\begin{aligned}Q_{ii} &= \frac{1}{\text{maximum acceptable value of } x_i^2} \\ R_{jj} &= \frac{1}{\text{maximum acceptable value of } u_j^2}\end{aligned}$$

11.4 Examples

12 Kalman Filter

12.1 Reference signals

Feed forward.

$$M \cdot N^{-1} = \tilde{M}$$

12.2 Pole assignment

$$A_{za} = A + BF + LC$$

$$C_{za} = -F$$

$$N = -(C_{cl}A_{cl}^{-1}\tilde{B}_{cl})^{-1}$$

$$M = \tilde{M}N$$

12.3 Random Values

Probability density function is used for determining the probability of a random variable falling within a particular range of values. This is done because a continuous random variable X often has zero probability of being one particular value.

$$Pr(a \leq X \leq b) = \int_a^b f_X(x)dx$$

The expected value (mean value) of a continuous random variable X can be determined from the probability density function $f_X(x)$ as follows:

$$E[X] = \int_{-\infty}^{\infty} x f_X(x)dx$$

The variance of X is defined as:

$$Var[X] = E[(X - E[X])^2]$$

For a multivariate random variable X, the expected value is a vector:

$$X = \begin{bmatrix} X_1 \\ X_2 \\ \vdots \\ X_n \end{bmatrix}$$

Covariance matrix

$$\Sigma = E[(X - E[X])(X - E[X])^T]$$

The variance for the different values can be found on the diagonal of the covariance matrix.

12.4 Kalman Filter

12.5 Examples