Projektkatalog

6. semester Signal Processing

2025

Indhold:

Projekterne kan enten anvendes direkte eller bruges som inspiration for jeres egne projektforlag.

To the english speaking: The project proposals can be used directly or as inspiration for you own proposals.
Some of the proposals are written in Danish. You will find machine translations of the text on a following page.

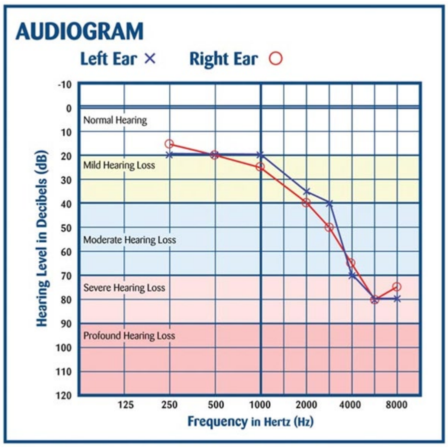
Det digitale høreapparat, ESD6 F25

Motivation

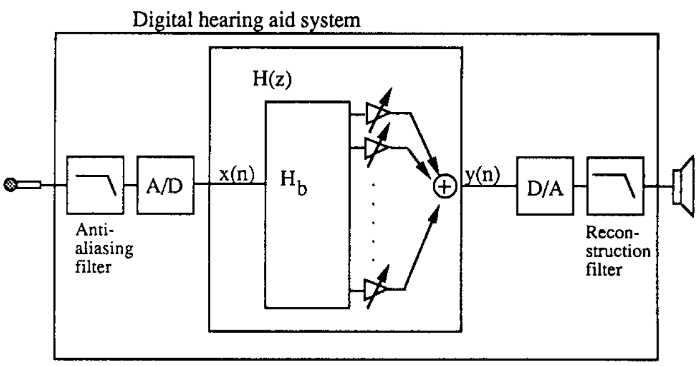
Høreapparater har siden umiddelbart efter 2. verdenskrig været genstand for en væsentlig del af den forskning, der er foregået indenfor elektronisk lyd-behandling. I de tidlige år og faktisk langt op i det 20. århundrede bestod et høreapparat primært af et simpelt elektronisk system, som over et bredt frekvens-område var i stand til at opfange lyd, og det vil primært sige tale, via en mikrofon, forstærke lyden op og gengive den i en hovedtelefon.

Et sådant apparat var selvsagt forbeholdt de få, som havde den økonomiske formåen til at anskaffe sig et sådant. Prisen til trods var den frembragte forbedring af taleforståeligheden dog ofte af tvivlsom karakter og i nogle tilfælde var brugen af et sådant apparat ligefrem direkte skadelig, idet lydniveauet var så højt, at den hørehæmmede kunne få yderligere høreskader – og det endda i den del af frekvensspektret, hvor brugeren havde en tilbageværende hørerest.

Det forholder sig nemlig sådan, at et høretab ikke nødvendigvis er lige-fordelt over hele det hørbare frekvensområde fra 20-20.000 Hz, men ofte kun i en del heraf. Derfor er der særlig god grund til kun at forstærke lyden i de frekvens-områder, hvor brugeren har et høretab. Nedenstående figur er et eksempel på et såkaldt audiogram, som viser det aktuelle høretab hos en patient set i forhold til "normal hørelse", hvilket indikeres ved $0dB$ Hearing Level.



Audiogrammet viser et typisk høretab, som er markant ved de høje(re) frekvenser, mens patienten har et svagt til moderat høretab ved lavere frekvenser. For om muligt at kompensere for et sådan høretab kunne man forestille sig et elektronisk apparat, som, helt overordnet set, har en frekvenskarakteristik fra indgang til udgang, der er den inverse af audiogrammets karakteristik. Det betyder, at signalet passerer u-forstærket igennem i det frekvensområde, hvor brugeren har normal hørelse, mens der i områder med høretab foretages en proportional forstærkning af signalet. Der er altså tale om en slags "equalizer", som forsøger at tilvejebringe en samlet frekvens-karakteristik, som er maksimal flad med et niveau omkring $0dB$. For at dette kan lade sig gøre, er det nødvendigt at inddele frekvensområde (i ovenstående eksempel fra $250Hz$ til $8kHz$) i en række individuelle frekvensbånd, på hvilke der så udføres den nødvendige forstærkning, hvorefter samtlige disse "sub-bands" så samles til et signal igen og sendes til høreapparatets højttaler. Opsplitningen i frekvensbånd foretages typisk (men ikke nødvendigvis udelukkende) ved at benytte en såkaldt Filter Bank (H_b), dvs. et antal båndpas-filtre, som arrangeres i en parallel konfiguration, hvor alle filtre arbejder på samme input signal. Dette betegnes "sub-band decomposition".



Med henblik på potentielt at forbedre taleforståeligheden yderligere, kunne en inspirerende ide være at udnytte den ofte ikke-ubetydelig høre-rest ved de lave(re) frekvenser til også at lytte til de høje(re) frekvenser. For at realisere en sådan feature skal informationen ved høje frekvenser flyttes ”nedad i frekvens” til det frekvensområde, hvor der altså fortsat er en eksisterende høreevne. En sådan funktion, som kan realiseres på en række forskellige måder, betegnes som ”Frequency Transformation”. Det må imidlertid forventes, at det resulterende talesignal bliver karakteriseret ved en væsentlig forvrængning grundet en sådan frekvenstransformation, men håbet er naturligvis, at brugeren, efter passende træning, får så meget yderligere information fra talesignalet, at den resulterende taleforståelighed forbedres.

Projekt-indhold

Projektet kunne naturligt tage udgangspunkt i en diskussion af det menneskelige øres anatomi og de fysiologiske årsager til typiske høretab, samt hvordan disse viser sig i relation til taleforståelighed. Dette vil give en grundlæggende forståelse af de egenskaber, som et høreapparat skal besidde, og de design-specifikationer, der skal opstilles for en eller flere dele af den samlede signalbehandlingsfunktion i et digitalt høreapparat, primært 1) sub-band baseret korrektion af lydniveau, og 2) flytning af information i frekvensdomænet.

Der kan i projektet med fordel studeres, analyseres og simuleres (C eller Matlab) flere forskellige metoder til såvel sub-band decomposition som frekvens-transformation, ligesom alternative nye metoder kan udvikles og sammenlignes med eksisterende algoritmer. I alle tilfælde vil det være naturligt at undersøge og realisere metoder til at tilvejebringe et spektrogram, dvs. visualisering af en tidsafhængig frekvensanalyse. Spektrogrammet kan benyttes til at undersøge og sammenligne, hvordan et signals effekt er fordelt i frekvens før og efter forskellige modifikationer.

En væsentlig del af projektet skal desuden fokusere på realtids-implementering af et eller flere delelementer af det samlede system. Der skal foretages implementering på en 16 bit fixed-point digital signalprocessor, hvilket fordrer, at de udvalgte algoritmer i første omgang underkastes undersøgelse af deres performance i endelig ordlængde. Dernæst ønskes tilvejebragt en prototype, som i realtid kan demonstrere virkningen af den/de udvalgte funktionalitet(er). Qua det givne applikationsområde kan DSP-koden undersøges mhp. at fastslå, hvorvidt den kan optimeres i form af reduktion af vitale performance-metrikker, såsom effekt- og hukommelsesforbrug samt eksekveringstid. En sådan analyse/optimering kan gennemføres med udgangspunkt i kode skrevet i enten C eller assembler, men kræver i begge tilfælde et detaljeret studie af processorens arkitektur, herunder både data- og kontrol-enheden.

Referencer

- H. Levitt, ”A Historical Perspective on Digital Hearing Aids: How Digital Technology has Changed Modern Hearing Aids”, Trends in Amplification, vol. 11, no. 1, March 2007, pp. 7-24.
- S.P. Philip et al., ”A Computationally Efficient 11 Band Non-Uniform Filter Bank for Hearing Aids Targeting Moderately Sloping Sensorineural Hearing Loss”, Journal of Microelectronics, Electronic Components and Materials, vol. 50, no. 3, 2020, pp. 153-167.
- A. Sokolova et al., “Multirate Audiometric Filter Bank for Hearing Aid Devices”, 55th Asilomar Conference on Signals, Systems, and Computers, 2021, pp. 1436-1442.
- Dava Smriga, “Frequency Lowering Fitting and Verification”, AudiologyOnline, https://www.audiologyonline.com/articles/frequency-lowering-fitting-and-verification-25644, Oct. 2019.
- A. Simpson, “Frequency-Lowering Devices for Managing High-Frequency Hearing Loss: A Review”, Trends in Amplification, vol. 13, no. 2, June 2009, pp. 87-106.

Forslagsstiller

Peter Koch, B4-206 (pk@es.aau.dk)

Translation:

The Digital Hearing Aid, ESD6 F25

Motivation

Hearing aids have been a significant focus of research in electronic sound processing since shortly after World War II. In the early years and well into the 20th century, a hearing aid primarily consisted of a simple electronic system capable of capturing sound, primarily speech, via a microphone, amplifying it, and reproducing it in a headphone. Such a device was reserved for the few who could afford it. Despite the cost, the improvement in speech intelligibility was often questionable, and in some cases, the use of such a device was directly harmful, as the sound level was so high that the hearing-impaired could suffer further hearing damage, even in the frequency range where the user had some residual hearing. Hearing loss is not necessarily evenly distributed across the audible frequency range from 20-20,000 Hz but often only in part of it. Therefore, it is particularly important to amplify sound only in the frequency ranges where the user has hearing loss.

The figure below is an example of an audiogram showing the current hearing loss of a patient compared to "normal hearing," indicated by 0 dB Hearing Level. The audiogram shows a typical hearing loss, which is significant at higher frequencies, while the patient has mild to moderate hearing loss at lower frequencies. To compensate for such hearing loss, one could imagine an electronic device with a frequency characteristic from input to output that is the inverse of the audiogram's characteristic. This means that the signal passes unamplified through the frequency range where the user has normal hearing, while in areas with hearing loss, the signal is proportionally amplified. This is a kind of "equalizer" that tries to provide an overall frequency characteristic that is maximally flat with a level around 0 dB. To achieve this, it is necessary to divide the frequency range (in the example from 250 Hz to 8 kHz) into several individual frequency bands, on which the necessary amplification is performed, after which all these "sub-bands" are combined into a signal again and sent to the hearing aid's speaker. The division into frequency bands is typically (but not necessarily exclusively) done using a so-called Filter Bank, i.e., a number of band-pass filters arranged in a parallel configuration, where all filters work on the same input signal. This is called "sub-band decomposition."

To potentially further improve speech intelligibility, an inspiring idea could be to use the often significant residual hearing at lower frequencies to also listen to higher frequencies. To realize such a feature, the information at high frequencies must be "shifted down in frequency" to the frequency range where there is still existing hearing ability. Such a function, which can be realized in several different ways, is called "Frequency Transformation." It is expected that the resulting speech signal will be characterized by significant distortion due to such frequency transformation, but the hope is that the user, after appropriate training, will get enough additional information from the speech signal to improve overall speech intelligibility.

Project Content

The project could naturally start with a discussion of the anatomy of the human ear and the physiological causes of typical hearing loss, as well as how these manifest in relation to speech intelligibility. This will provide a fundamental understanding of the characteristics that a hearing aid must possess and the design specifications that must be established for one or more parts of the overall signal processing function in a digital hearing aid, primarily 1) sub-band based correction of sound level, and 2) shifting of information in the frequency domain. The project can advantageously study, analyze, and simulate (in C or Matlab) several different methods for both sub-band decomposition and frequency transformation, as well as develop and compare new methods with existing algorithms. In all cases, it will be natural to investigate and realize methods to provide a spectrogram, i.e., visualization of a time-dependent frequency analysis. The spectrogram can be used to investigate and compare how a signal's power is distributed in frequency before and after various modifications. A significant part of the project should also focus on real-time implementation of one or more elements of the overall system. Implementation on a 16-bit fixed-point digital signal processor is required, which necessitates that the selected algorithms are first examined for their performance in final word length. Then, a prototype should be provided that can demonstrate the effect of the selected functionality in real-time. Given the application area, the DSP code can be examined to determine whether it can be optimized in terms of reducing vital performance metrics, such as power and memory consumption and execution time. Such an analysis/optimization can be carried out based on code written in either C or assembler but requires a detailed study of the processor's architecture, including both the data and control unit.

References

- H. Levitt, "A Historical Perspective on Digital Hearing Aids: How Digital Technology has Changed Modern Hearing Aids," Trends in Amplification, vol. 11, no. 1, March 2007, pp. 7-24.
- S.P. Philip et al., "A Computationally Efficient 11 Band Non-Uniform Filter Bank for Hearing Aids Targeting Moderately Sloping Sensorineural Hearing Loss," Journal of Microelectronics, Electronic Components and Materials, vol. 50, no. 3, 2020, pp. 153-167.
- A. Sokolova et al., "Multirate Audiometric Filter Bank for Hearing Aid Devices," 55th Asilomar Conference on Signals, Systems, and Computers, 2021, pp. 1436-1442.
- Dava Smriga, "Frequency Lowering Fitting and Verification," AudiologyOnline, https://www.audiologyonline.com/articles/frequency-lowering-fitting-and-verification-25644, Oct. 2019.
- A. Simpson, "Frequency-Lowering Devices for Managing High-Frequency Hearing Loss: A Review," Trends in Amplification, vol. 13, no. 2, June 2009, pp. 87-106.

Proposer

Peter Koch, B4-206 (pk@es.aau.dk)

Kilde: Steinway & Sons

Klaver tuner

Baggrund:

En tuner er et måleinstrument, der kan hjælpe ved stemning (tuning) af musikinstrumenter.

Guitar tuner. Kilde: www.yamaha.dk

Strengeinstrumenter skal jævnligt stemmes, da de går ud af stemning bl.a. på grund af svingende temperatur og fugtighed i omgivelserne. Når man stemmer et strengeinstrument spænder eller løsner man strengen indtil den svinger med den korrekte frekvens – altså den rigtige tone. Dette er en relativ triviel opgave for instrumenter, hvor tonerne laves af enkeltstrenge, som f.eks. guitar, bas og violin. Der er straks en meget mere kompleks opgave når man har at gøre med et klaver, idet de enkelte toner på klaveret dannes af henholdsvis 1, 2 eller 3 strenge.

Når klaverstemmeren skal stemme en tone, som dannes af 3 strenge bruges nogle gummikiler og filtbånd, som sættes mellem strengene for at dæmpe svingningerne, så kun én af de 3 strenge får lov til at svinge og dermed afgive lyd. Kilerne flyttes herefter så en anden streng får lov at svinge, og til sidst rettes den sidste streng ind, så den stemmer overens med de to andre. Dette er en langsommelig proces, da der kan være op til omkring 230 strenge på et klaver, og det ville være en stor fordel hvis dæmpningen af nabostrenge kunne undgås.

Projektet:

Som den centrale del, skal der designes og konstrueres en klaver tuner, som kan give klaverstemmeren et overblik over frekvenserne af flere samtidigt svingende strenge for en tone på klaveret baseret på input fra en mikrofon (eller et accelerometer) med tilhørende forstærker og efterfølgende A/D converter. Det er ikke nogen triviel opgave, da frekvenserne ligger meget tæt på hinanden, hvorfor der vil være behov for at benytte specielle signalbehandlingsalgoritmer til opgaven.

Tuneren kan enten realiseres som et (håndholdt) måleapparat til brug for klaverstemmeren under selve stemningen af klaveret, eller som et IOT device, som installeres i klaverer og løbende kan informere om kvaliteten af klaverets stemning og sende bud efter klaverstemmeren når det er tid for servicering. Dette kunne f.eks være meget relevant for musikskoler, koncerthuse o. lign., som har en større samling af klaverer.

Forslagsstiller: Flemming Christensen (fc@es.aau.dk)

Translation:

Piano Tuner

Background: A tuner is a measuring instrument that can assist in tuning musical instruments. String instruments need to be tuned regularly as they go out of tune due to fluctuating temperature and humidity in the surroundings. When tuning a string instrument, the string is tightened or loosened until it vibrates at the correct frequency – the right tone. This is a relatively trivial task for instruments where the tones are made by single strings, such as guitar, bass, and violin. It becomes a much more complex task when dealing with a piano, as the individual tones on the piano are produced by 1, 2, or 3 strings. When the piano tuner tunes a tone produced by 3 strings, rubber wedges and felt strips are used to dampen the vibrations, so only one of the 3 strings is allowed to vibrate and thus produce sound. The wedges are then moved so another string is allowed to vibrate, and finally, the last string is adjusted to match the other two. This is a slow process, as there can be up to around 230 strings on a piano, and it would be a great advantage if the damping of neighboring strings could be avoided.

The Project: The central part of the project is to design and construct a piano tuner that can give the piano tuner an overview of the frequencies of several simultaneously vibrating strings for a tone on the piano based on input from a microphone (or an accelerometer) with an associated amplifier and subsequent A/D converter. This is not a trivial task, as the frequencies are very close to each other, requiring the use of special signal processing algorithms. The tuner can either be realized as a (handheld) measuring device for use by the piano tuner during the tuning of the piano or as an IoT device installed in pianos that can continuously inform about the quality of the piano's tuning and send for the piano tuner when it is time for servicing. This could be very relevant for music schools, concert halls, etc., which have a larger collection of pianos.

Proposer: Flemming Christensen (fc@es.aau.dk)

Sound Analyzer for Live Concerts

The sound ”engineer” is often an underpaid guy, that loves music, but does not necessary have much knowledge in the field and over time also a degraded hearing, due to too high sound pressure level at live concerts. He normally only have his own ears and taste to control the complicated process of producing a high quality experience for the audience.
(It must be complicated seen in the perspective of how often it goes wrong!)

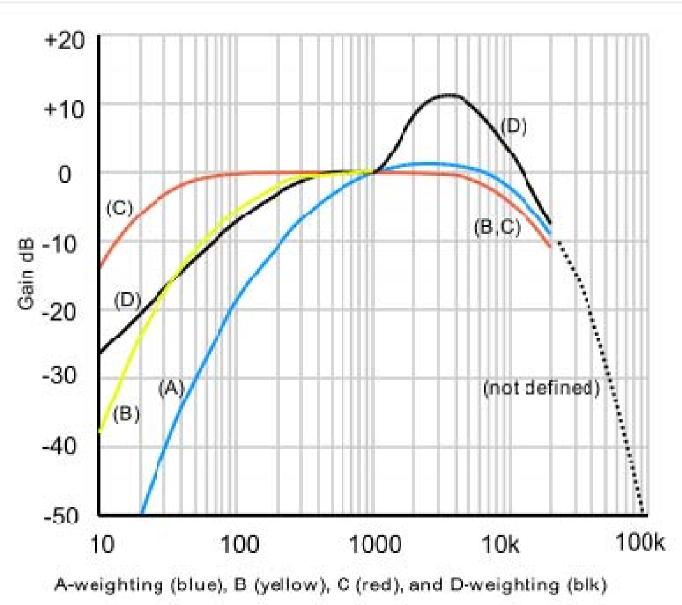
The sound engineer must control the mixing/gain of many channels which include compressing, equalizing, stereo panning, ending up with setting the main output level (sound pressure level), which normally is far too high, if quality is wanted, with the risk of hearing damage among the audience or the people working at the venue.

To support the sound engineer and the audience, an analyzer would be a big help. Equalizing the sound might often be made using a 1/3 octave equalizer, and for that reason a similar analyzer would be a big help, as being able to visualize the frequency spectre, to find the right filter to tune, not relying only on the hearing. Such an analyzer can be made of 30 high order band pass filters and level meters which is a lot of signal processing going on in real time, but it can be made with only 3 BP filters, and successive down-sampling, using multi rate signal processing, and thereby saving a lot of signal processing power.

The overall sound doses is also wanted, doses is an integration of sound pressure level over time, and should not exceed 80 dBA in 8 hours over a day. (85 dBA is the limit at work places)

There are standards for sound level meters and analyzers that can/should be used: IEC-1260

Sofus Birkedal Nielsen



Sound Level Meter

Standard frequency weighting curves

Sound Dose = Sound Energy integrated over time

Max Sound dose at work-places: 85 or 80 dBa in 8 hours meaning:

Constant Level dBA		Time:
85	80	8 Hours
88	83	4 –
91	86	2 -
93	89	1 –
97	92	15 min
100	95	7,5 min
103	98	3,7 min
106	101	2 min

Sound Effect Unit for Guitar or Vocalist.

An electric guitar is an instrument that in itself does not have an interesting sound, as the electrical guitar does not have a resonating body as a classical guitar. The sound is picked up from one or more electromagnetic pickup under the strings, and can be amplified and processed in various ways. Amplification is not enough to make the guitar sound good the signal also has to be processed or changed, and in that sense both analog and digital signal processing can be used. These effect units were in beginning made with a single analog effect, and known as guitar pedals. With the introduction of digital signal processing and signal processors many new and interesting effects has appeared and only the fantasy seems to be the limit of new effects. There are at lot of well known 'standard' effect such as:

Chorus
Flanger
Delay
Reverberation
Overdrive
Equalising
Wav-Wav
etc.

Some of these effects can best be implemented using analog electronics such as the wav-wav effect, and distortion, while others effects can better or only be implemented using digital signal processing such as reverb units, flanger and chorus while other effects such as equalizing can be implemented in both analog and digital signal processing.

The input signal is an analog signal coming from an electrical guitar needing a suitable preamplifier before the effect units. The effect unit is implemented at a DSP development system including A/D and D/A converters. A number of effects is chosen and implemented as software at the DSP system. It is obvious that the system has to work in real time, so the implementation has to be code efficient in order to have more effects running at the same time, with sufficient sound quality.

User interface is also needed, but might be kept at a low level.

An electrical guitar and a guitar amplifier is available for the project, and you do not need to be able to play electrical guitar to make this project.

Sound Effect for Vocal
Instead of an effect unit for guitar, the focus could also be an effect unit for vocalist, with effects as compressor, chorus, harmony, de-esser, reverb, equalizing.

Inspiration can be found at www.tcelectronic.com

Sofus Birkedal Nielsen

Effect units from TC-Electronics for VOCAL

Guitar Effect Units

SP6 projektforslag:

Maskin-overvågning for at minimere breakdown.

Maskiner slides over tid, og det kan være svært at forudsige hvornår en reparation er reparation er nødvendig. Kunsten er at optimere en reparation, så det er så lidt generende som muligt, på den anden side er det for dyrt at udskifte eller reparere før det er nødvendigt (”if it arnt broken – dont fix it”).

Maskiner er mange ting så som:

- roterende maskiner (motorer, generatorer, rullende…. Alle med lejer (kugle lejer eller rulle-lejer)
- translatoriske eller drejende maskindele (robotter)
- Hydrauliske maskiner
- Rystende maskiner
- Kedelanlæg
- …

Alle bevægelige og roterende maskindele slides afhængig af hvor meget de belastes.

Indgår maskinen i en produktionskæde kan et pludselig breakdown betyde store økonomiske tab både for fabriksejer og personale, som måske må sendes hjem i en periode, så en måske relativ billig reparation kan få store økonomiske følger. Så det drejer sig om at være forberedt, og vide hvornår det er tid til en kontrolleret udskiftning før maskinen fejler. At skifte en gearkasse i en offshore vindmølle kan løbe op i millioner af kroner, fra pålidelig kilde forlyder det, at når man har lejer og sejlet en kran ud på vandet og skiftet en gearkasse er der ½ mill. Kr at spare, hvis man samtidig kan udføre en lignende reparation på en af nabomøllerne hvis man kan se at den også er slidt, frem for at vente til et senere nedbrud.

For at kunne se f.eks. en roterende maskine trænger til en reparation kan man med passende mellemrum se om der sker en forandring i vibrationer eller lyd enten niveaumæssigt eller frekvensmæssingt, her bruges ofte et accellerometer til at opsamle vibrationer fra et bestemt punkt, ofte et kuleleje. (på samme måde som lægen bruger et stetoskop). En frekvensanalyse over tid kan så afsløre om der er fare for et nedbrud.

Der tænkes her på at udvikle et håndholdt analyseudstyr evt. med grafisk display som kan vise og sammenligne et nyt og et gammelt spekter og evt. selv afgøre om der er fare på færde.

Man kunne også tænke på andre anvendelser, så som et autonomt system der via sms kan give besked om en ny måling eller slå alarm når det er ved at være kritisk.

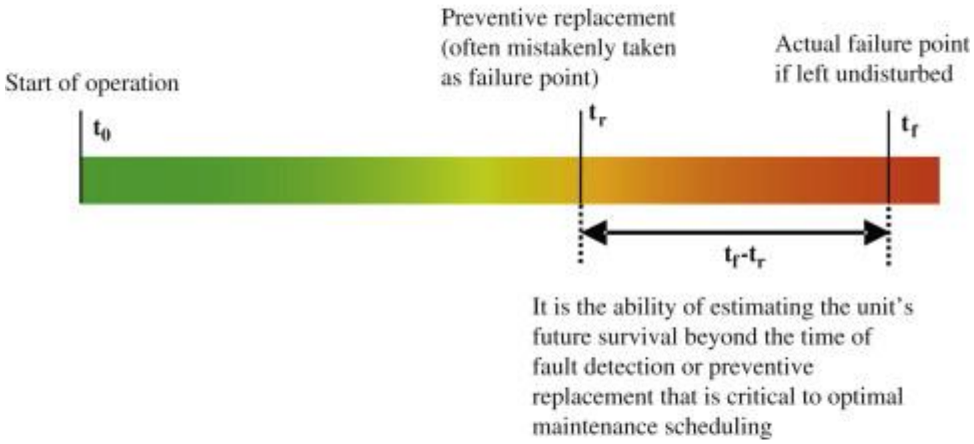
Der er mulighed for mange applikationer.

Systemet baseres på en 16/32 bit fixed point DSP med lavt strømforbrug.

Der kan findes meget litteratur om emnet, så som:

http://www.sciencedirect.com/science/article/pii/S0888327005001512
http://www.bkvibro.com/products/portable-instruments/vibroport-80.html
http://www.bksv.com/Library/Technical%20Reviews?year=1989-1985&st=1989-1985

Sofus Birkedal Nielsen



Angiver en tidsline for udviklingen af en slidtage og hvornår man bør sætte ind, se:

http://www.sciencedirect.com/science/article/pii/S0888327005001512

Eksempel på et håndholdt vibrations-udstyr.

http://www.bkvibro.com/products/portable-instruments/vibroport-80.html

Translation:

Project Proposal: Machine Monitoring to Minimize Breakdowns

Machines wear out over time, and it can be difficult to predict when a repair is necessary. The art is to optimize a repair so that it is as little disruptive as possible. On the other hand, it is too expensive to replace or repair before it is necessary ("if it ain't broken – don't fix it").

Machines include many things such as:

- Rotating machines (motors, generators, rolling… all with bearings (ball bearings or roller bearings))
- Translational or rotating machine parts (robots)
- Hydraulic machines
- Vibrating machines
- Boiler systems

All moving and rotating machine parts wear out depending on how much they are loaded. If the machine is part of a production chain, a sudden breakdown can mean significant financial losses for both the factory owner and staff, who may have to be sent home for a period. Thus, a relatively cheap repair can have major economic consequences. Therefore, it is about being prepared and knowing when it is time for a controlled replacement before the machine fails.

Replacing a gearbox in an offshore wind turbine can cost millions of kroner. Reliable sources indicate that when bearings and a crane are sent out to sea to replace a gearbox, there is half a million kroner to save if a similar repair can be performed on a neighboring turbine if it is also worn, rather than waiting for a later breakdown.

To see, for example, if a rotating machine needs a repair, one can periodically check for changes in vibrations or sound, either in terms of level or frequency. An accelerometer is often used to collect vibrations from a specific point, often a ball bearing (similar to how a doctor uses a stethoscope). A frequency analysis over time can then reveal if there is a risk of a breakdown.

The idea is to develop a handheld analysis device, possibly with a graphical display, that can show and compare a new and an old spectrum and possibly determine if there is a danger. One could also think of other applications, such as an autonomous system that can send an SMS notification of a new measurement or sound an alarm when it is critical. There are many application possibilities.

The system is based on a 16/32 bit fixed-point DSP with low power consumption. There is much literature on the subject, such as:

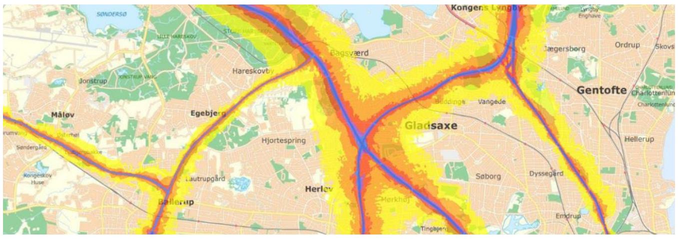
- ScienceDirect Article
- Vibroport 80
- Technical Reviews

Sofus Birkedal Nielsen

Målestation til indsamling af miljødata

Baggrund:

Der er et stadigt stigende fokus på - og interesse for - løsninger, der kan hjælpe til at forbedre miljøet. For at kunne agere korrekt i den sammenhæng, har myndighederne behov for det bedst mulige vidensgrundlag herunder hvad f.eks. støjniveauet reelt er forskellige steder i miljøet omkring os. P.t. bruges modeller for køretøjers støjemission samt modeller for lydudbredelse til at kortlægge den gennemsnitlige trafikstøjpåvirkning på givne adresser i Danmark (se: https://miljoegis.mim.dk/spatialmap?profile=noise)



Der er få begyndende tiltag til at indsamle og visualiserer data i real-tid. Se f.eks.:

https://ftdynamicnoisemapv2test.azurewebsites.net/site/nyborg

De ovenstående inkluderer dog kun – eller væsentligst – trafikstøj og ikke påvirkningen fra andre støjkilder.
I forbindelse med dataopsamling, kunne man fristes til at lave en overvågning af støjen ved hjælp af optagelse på forskellige lokationer og efterfølgende behandling centralt (cloud-baseret). Dette er dog ikke hensigtsmæssigt, da det vil betyde mulig overvågning af samtaler m.v. med dertil hørende etiske problemer (og mulig overtrædelse af GDPR). En løsning bør derfor måle relevante fysiske størrelser, som ikke i sig selv afslører personfølsomt indhold. Dette kunne f.eks. være A-vægtet lydtrykniveau midlet efter de gældende standarder, eller tidsmidlet oktavbåndsfiltreret lydtryk.

Indhold:

Der skal opbygges hardware til måling, lagring og transmission af miljødata – i dette tilfælde støj.

Systemet skal kunne foretage lydmåling med relevante behandlinger af lydsignalet f.eks. A-vægtning, oktavbåndsfiltrering, tidsmidling m.v. ihht. gældende regler og standarder. Data skal kunne overføres trådløst til en central database (optional).

Som et ekstra element kunne inddrages muligheden for batteridrift med heraf følgende krav til lavt strømforbrug og udnyttelse af diverse sleep-modes i HW.

Forslagsstiller:

Flemming Christensen (fc@es.aau.dk)

Translation:

Measurement station for collecting environmental data

Background: There is an increasing focus on - and interest in - solutions that can help improve the environment. In order to act correctly in this context, the authorities need the best possible knowledge base, including what, for example, the noise level actually is in different places in the environment around us. Currently, models for vehicle noise emissions and models for sound propagation are used to map the average traffic noise impact at given addresses in Denmark (see: link)). There are a few initial steps to collect and visualize data in real-time. See e.g.: link. However, the above only includes - or mainly - traffic noise and not the impact from other noise sources.

In connection with data collection, one might be tempted to monitor the noise by recording at different locations and subsequently processing centrally (cloud-based). However, this is not appropriate, as it would mean possible monitoring of conversations etc. with associated ethical problems (and possible violation of GDPR). A solution should therefore measure relevant physical quantities that do not in themselves reveal sensitive personal content. This could, for example, be A-weighted sound pressure level averaged according to the applicable standards, or time-averaged octave band filtered sound pressure.

Content: Hardware must be built for measuring, storing and transmitting environmental data - in this case noise. The system must be able to perform sound measurement with relevant processing of the sound signal e.g. A-weighting, octave band filtering, time averaging etc. according to applicable rules and standards. Data must be able to be transferred wirelessly to a central database (optional). As an additional element, the possibility of battery operation could be included with the resulting requirements for low power consumption and utilization of various sleep modes in HW.

Proposer: Flemming Christensen (fc@es.aau.dk)

Målesystem til bestemmelse af rumakustiske materialeparametre.

Baggrund:

Akustikken i rum kan udtrykkes ved hjælp af parametre som f.eks. efterklangstiden. Efterklangstiden i et rum udtrykker hvor hurtigt lydfeltet i rummet dæmpes som funktion af tiden, og den giver et mål for den umiddelbare hørbare forskel på den akustiske kvalitet af forskellige rum. Tænk f.eks. på et klap i et badeværelse eller i en stor kirke. Det er rummets volumen i kombination med materialerne på rummets overflader, der er afgørende for hvor lang en efterklangstid et rum har.

Når man skal designe nye rum til forskellige formål, beregner akustikingeniører typisk en forventet efterklangstid af et rum baseret på materialeparametre for overfladematerialerne.

Et materiales absorptionsegenskaber kan grundlæggende bestemmes på to forskellige måder: Ved måling i et efterklangsrum hvilket er meget ressource og pladskrævende eller ved måling i et såkaldt impedansrør [1,2]. I begge tilfælde måles impulsresponser for materialet, hvorudfra frekvensafhængige absorptionskoefficienter kan bestemmes.

Dette projekt omhandler konstruktion af et impedansrør med tilhørende impulsrespons-målesystem til bestemmelse af byggematerialers absorptionskoefficienter. (I laboratoriet findes et ældre system, der kan anvendes til verifikation af det udviklede).

Indhold:

- Litteraturstudie om rumakustik, rumakustiske parametre og materialeegenskaber.
- Valg af mikrofoner og signalbehandlingshardware
- Udvikling og implementering af målealgoritme baseret på nedenstående reference 2.
- Implementering af et funktionelt system

Referencer:

1) DS/EN ISO 10534-1:2001 Akustik – Bestemmelse af lydabsorptionskoefficient og impedans i impedansrør – Del 1: Metode baseret på måling af standbølgeforhold
2) DS/EN ISO 10534-2:2023 Akustik – Bestemmelse af akustiske egenskaber i impedansrør – Del 2: Lydabsorptionskoefficient og overfladeimpedans ved normalt lydindfald udført med to mikrofoner

Forslagsstiller: Flemming Christensen (fc@es.aau.dk)

Målesystem med impedansrør fra Brüel & Kjær

Efterklangsrum

Translation:

Measurement system for determining room acoustic material parameters.

Background: The acoustics in a room can be expressed using parameters such as reverberation time. The reverberation time in a room indicates how quickly the sound field in the room decays as a function of time, providing a measure of the immediate audible difference in the acoustic quality of different rooms. Think, for example, of a clap in a bathroom or in a large church. It is the volume of the room in combination with the materials on the room's surfaces that determines how long a reverberation time a room has. When designing new rooms for various purposes, acoustic engineers typically calculate an expected reverberation time of a room based on material parameters for the surface materials. The absorption properties of a material can fundamentally be determined in two different ways: by measurement in a reverberation room, which is very resource and space-intensive, or by measurement in a so-called impedance tube [1,2]. In both cases, impulse responses for the material are measured, from which frequency-dependent absorption coefficients can be determined. This project involves the construction of an impedance tube with an associated impulse response measurement system to determine the absorption coefficients of building materials. (In the laboratory, there is an older system that can be used to verify the developed system).

Content:

- Literature study on room acoustics, room acoustic parameters, and material properties.
- Selection of microphones and signal processing hardware.
- Development and implementation of a measurement algorithm based on the reference below.
- Implementation of a functional system.

References:

1) DS/EN ISO 10534-1:2001 Acoustics – Determination of sound absorption coefficient and impedance in impedance tubes – Part 1: Method based on standing wave ratio measurement.

2) DS/EN ISO 10534-2:2023 Acoustics – Determination of acoustic properties in impedance tubes – Part 2: Sound absorption coefficient and surface impedance with normal sound incidence performed with two microphones.

Proposer: Flemming Christensen (fc@es.aau.dk)