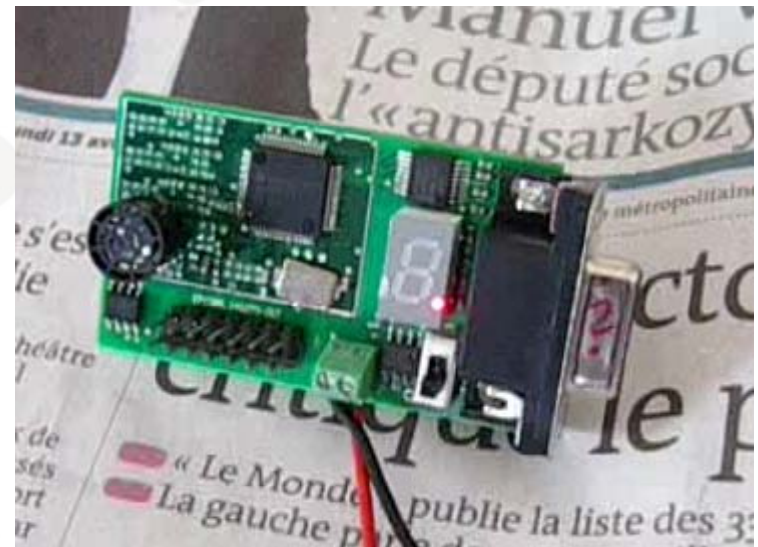


7.3 Samplerate-konvertering

- Ultralyd kommunikasjonssystem for kombinasjon med RFID
 - Signal $40 \text{ kHz} \pm 2 - 4 \text{ kHz} \Leftrightarrow B = 4 - 8 \text{ kHz}$
 - Anti-aliasing filter \Leftrightarrow transducers båndpasskarakteristikk
 - $f_s \approx 17.7 \text{ kHz}$: Båndpass sampling
 - Filtrerer og desimerer med 8
 - $f_{s2} = 17.7/8 \approx 2.2 \text{ kHz}$
 - Videre prosessering for deteksjon og demodulasjon
 - S. Holm, "Hybrid ultrasound-RFID indoor positioning: Combining the best of both worlds," IEEE Int. Conf. RFID, Orlando, FL, April, 2009.

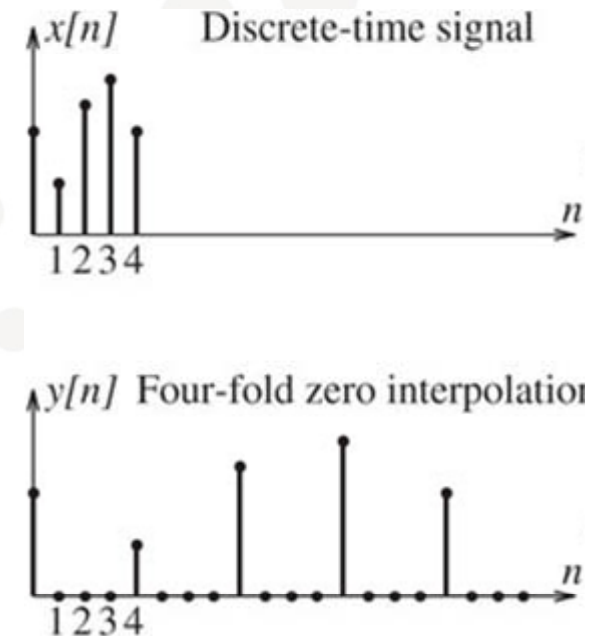


Ned- og oppsampling i mp2/mp3

- Mp2: Deler i 32 bånd
- Datarater i koderen (mono)
 - Inn: 44100 samples pr sek
 - Ut av filterbank: 44100×32
 - Men hver filterbank har bare $1/32$ båndbredde => desimering: $44100/32 \times 32 = 44100$
 - Så gjøres reduksjon av antall bit ut fra persepsjonskriterier
- Tilsvarende oppsampling i dekodeeren

Samplerate-konvertering

- Behov for å konvertere uten å gå om analogt domene
- Oppsampler ved å sette inn 0-er:
 - $y[n]=x\uparrow[n/N] \neq 0$ for $n=kN$
 - Signalet strekkes i tid



$$Y_p(F) = \sum y[n]e^{-j2\pi nF} = \sum y[kN]e^{-j2\pi kNF} = \sum x[k]e^{-j2\pi kNF} = X_p(NF)$$

- Altså skaleres spektret (komprimeres)
- Lenger i tid \Leftrightarrow kortere i frekvens

Oppsampling: Spektrumskompresjon

- Kompresjon av spektret ved oppsampling

$$Y_p(F) = X_p(NF)$$

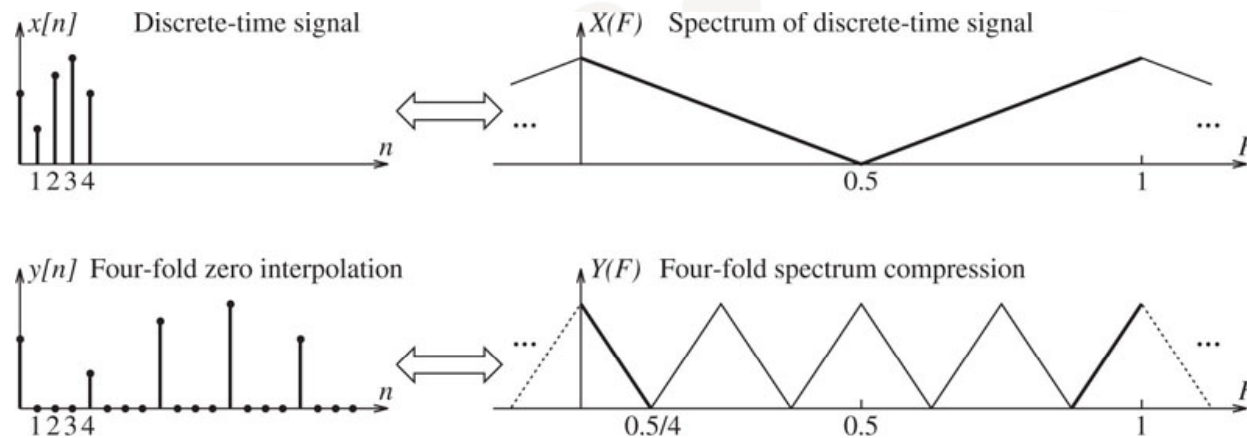


FIGURE 7.16 Zero interpolation of a signal leads to spectrum compression. Zero interpolation of the signal $x[n]$ by a factor of 4 results in the interpolated signal $y[n]$. The spectrum $Y(F)$ shows four-fold compression compared to the spectrum $X(F)$.

Oppsampling: Anti-imaging filter

- Sett inn $N-1$ nuller mellom samplene
- \Rightarrow komprimert spektrum; også replikaer som må filtreres bort ($\Leftrightarrow (N-1)/N$ av energien fjernes)
- Anti-imaging filter med kutt ved $0.5/N$, Gain N

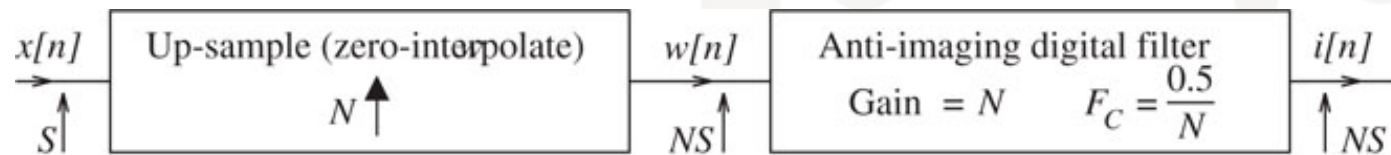


FIGURE 7.17 Sampling rate increase by an integer factor N requires zero interpolation followed by lowpass filtering. Zero interpolation results in N -fold compression of the spectrum. The central period contains N compressed images. All but one is filtered out by the anti-imaging filter with cutoff frequency $F_c = 0.5/N$. This produces a signal sampled at N times the original rate

Oppsampling

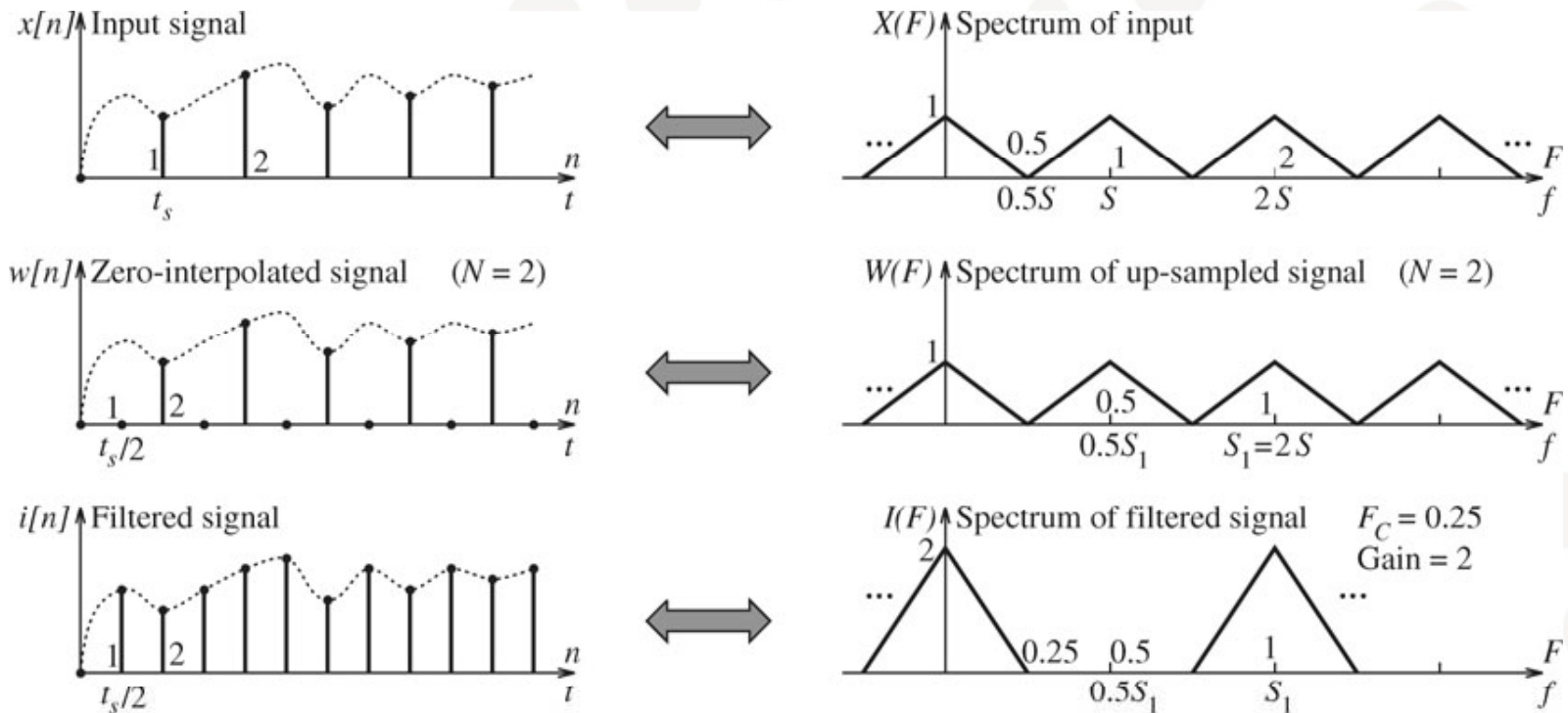


FIGURE 7.18 Spectra of signals when increasing the sampling rate by two. The original spectrum has one image per period. Zero interpolation produces two compressed images per period. A lowpass filter with a cutoff frequency of $F_C = 0.25$ and a gain of 2 eliminates one image. Filtering of the spurious image in the spectrum is equivalent to replacing the zeros in the zero-interpolated signal by actual values of the original signal. The signal $i[n]$ is thus sampled at twice the original rate

Nedsampling: Spektrumsstrekking

- Ved nedsampling strekkes spektret
- Desimert signal $y[n]=x[nM]$ – komprimeres i tid
- Nedsampling med faktor M

$$Y_p(F) = (1/M)X_p(F/M)$$

- Må ha med $1/M$ for å bevare energien
- Må først sørge for båndbegrensning til $|F|<0.5/M$

Nedsampling: Anti-aliasing filter

- Digital lavpass, kutt ved $=0.5/M$
- Kast $M-1$ samples mellom hver sample

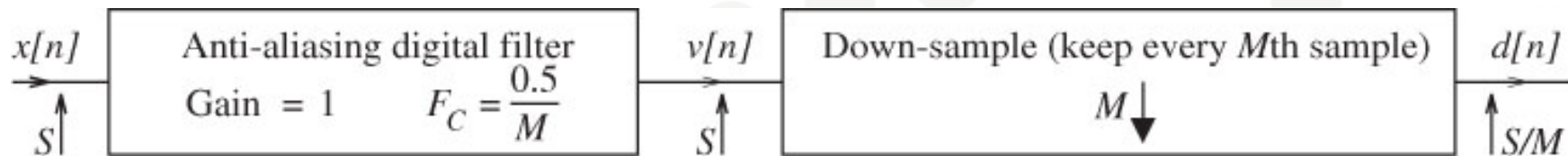
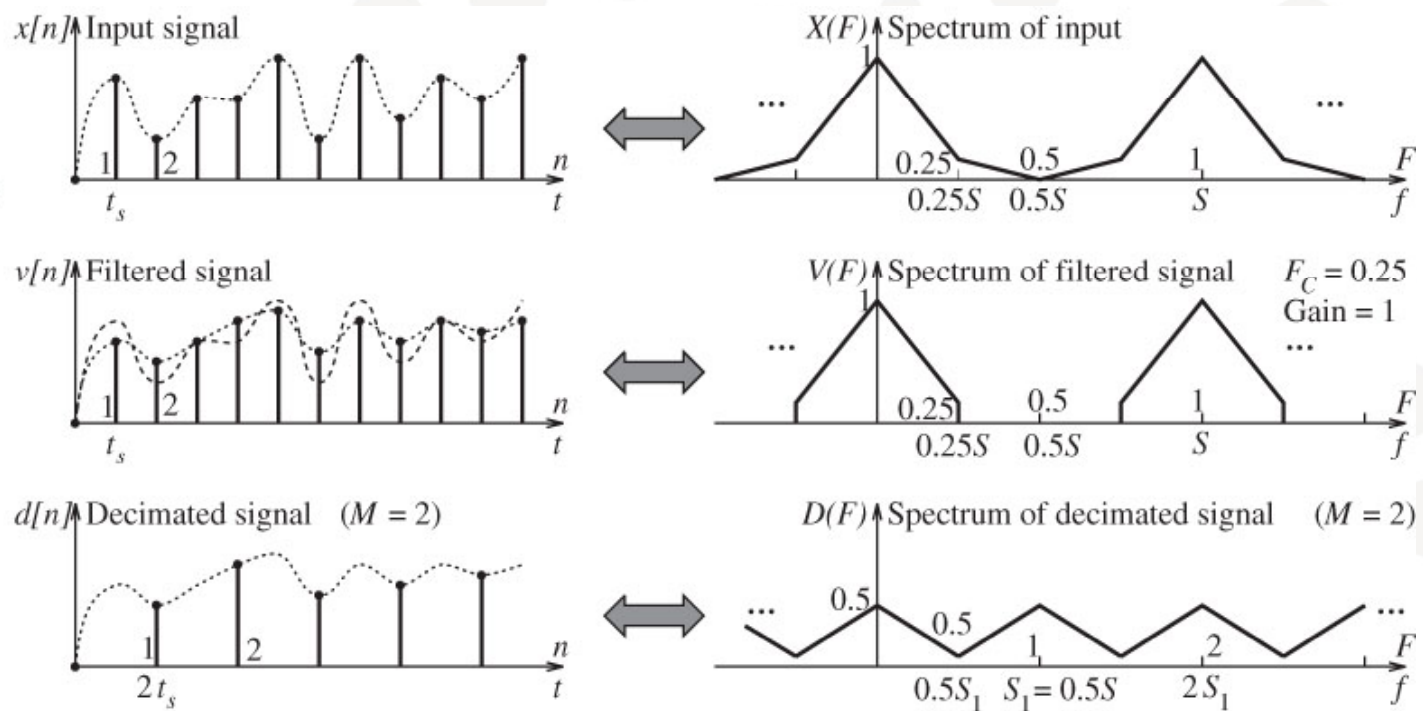


FIGURE 7.19 Sampling rate reduction by an integer factor M requires a lowpass filter followed by decimation (down-sampling). The lowpass filter bandlimits the signal to $F = 0.5/M$. Decimation by M stretches the spectrum and produces a signal sampled at $1/M$ times the original rate

Nedsampling

FIGURE 7.20 The spectra of various signals during sampling rate reduction by two. A lowpass filter with a cutoff frequency of $F_C = 0.25$ bandlimits the spectrum. Decimation by 2 stretches the spectrum. The signal $d[n]$ is thus sampled at half the original rate



Opp/ned-sampling

- N/M
- Husk! Alltid oppsample først, her med N
Så nedsample med M
- De to lavpassfilterne slås sammen til ett:
 - Gain $1/N$
 - Cut-off: den minste av $0.5/M$ og $0.5/N$

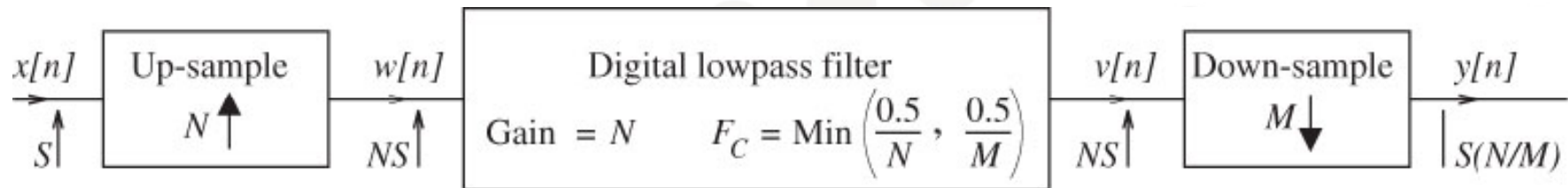


FIGURE 7.21 Illustrating a sampling-rate change by M/N . The first step is up-sampling by N . The second step is lowpass filtering using a gain of N and a cutoff frequency that is the smaller of $0.5/M$ and $0.5/N$. The final step is down-sampling by M

7.4 Kvantisering

- Uniform kvantisering: trinnene er like
 - B bits $\Rightarrow L=2^B$ trinn; I praksis vanlig med $B=8, 12, 16, 24$
- Avrunding, avkorting (trunkering), fortegns-avkorting

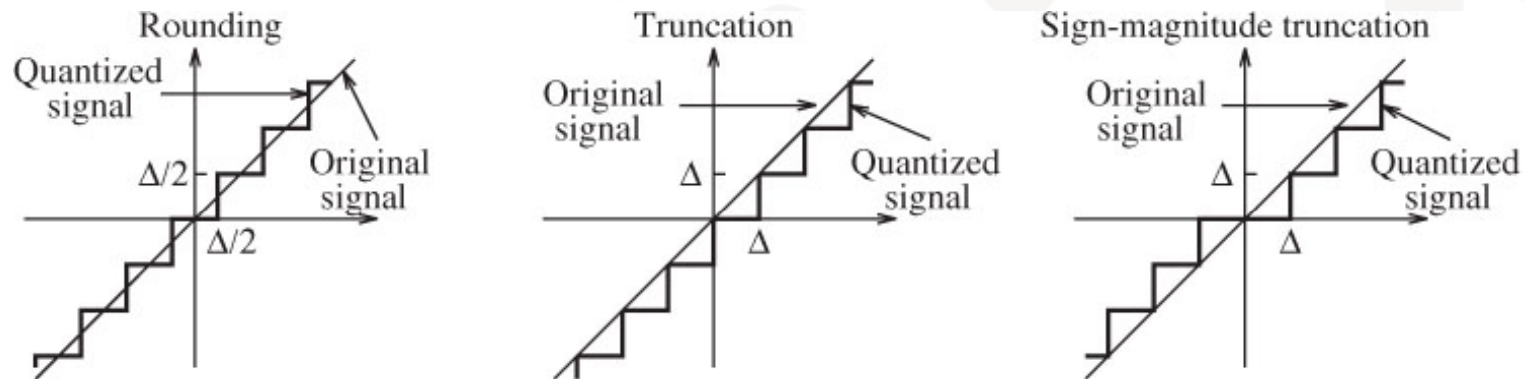


FIGURE 7.22 Various ways of quantizing a signal. The quantized value is chosen as the nearest quantization level when rounding or the next lower level when truncating. In sign-magnitude truncation, the absolute value (magnitude) is truncated and the actual sign is restored afterwards

Kvantisering

- Hva skjer hvis inngangsverdi er over maks-verdien?
- Klipping
 - Metning eller nulling

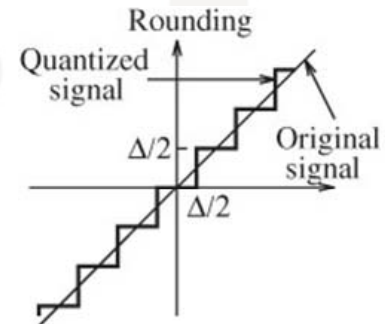
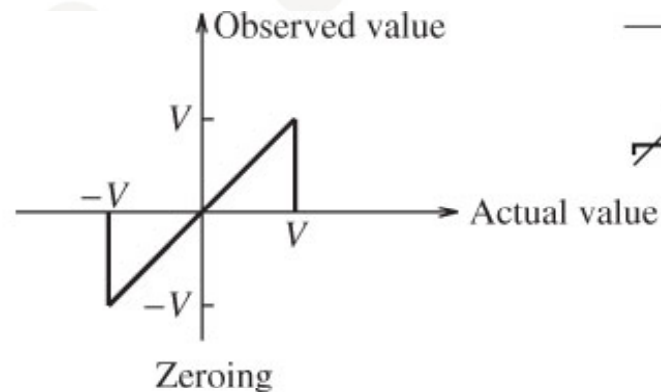
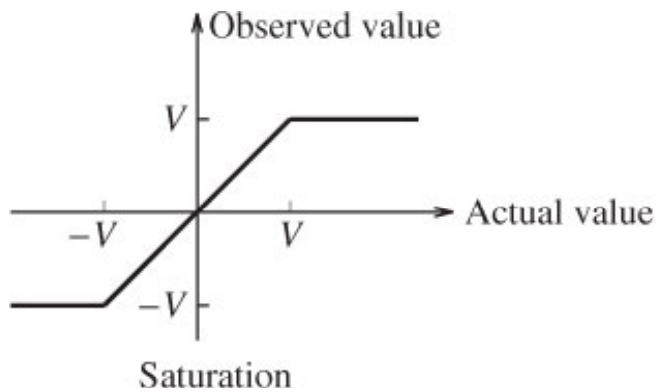
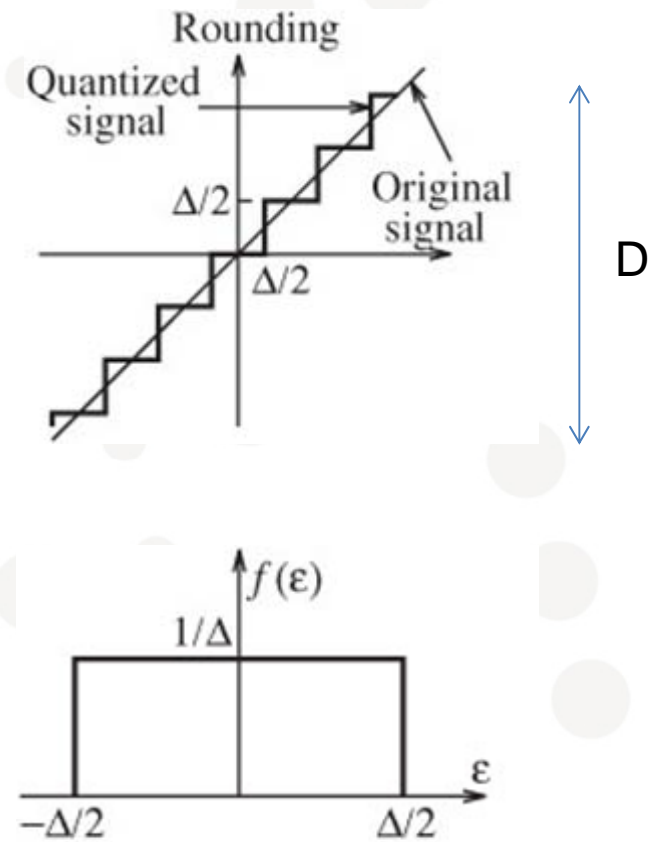


FIGURE 7.23 Two overflow characteristics. In saturation, values outside the full-scale range are set to the full-scale value itself. In zeroing, values outside the full-scale range are set to zero

- Demo: kvantisering.m

Kvantiseringfeil - kvantiseringstøy

- Feil $\varepsilon = x[n] - x_Q[n]$
- Maks dynamikk $D = X_{\max} - X_{\min}$
- Kvantiseringstrinn $\Delta = D/L = D/2^B$
- Dynamikk $\approx D/\Delta = 2^B$
 - dB: $20\log(2^B) = 20\log 2 \cdot B = 6 \cdot B$
 - CD 16 bit \Rightarrow 96 dB dynamikk
 - Mer nøyaktig beregning følger
- Feil, statistisk: $-\Delta/2 < \varepsilon < \Delta/2$
- B stor nok \Rightarrow feil uniformt fordelt, + ukorrelert med signal
- Bruker dither for å dekorrelere feil



Kvantiseringssfeilen

- Effekt i kvantiseringssstøyen
- pdf: $f(\epsilon) = 1/\Delta$ for $|\epsilon| < \Delta/2$
- Støyeffekt = variansen:

$$P_N = \sigma^2 = \int_{-\infty}^{\infty} \epsilon^2 f(\epsilon) d\epsilon = \frac{1}{\Delta} \int_{-\Delta/2}^{\Delta/2} \epsilon^2 d\epsilon = \frac{\Delta^2}{12}$$

- $\Delta = D/L$:
- $10\log(P_N) = 10\log(D^2/12L^2) = 20\log D - 20\log L - 10.8$

Signal-støy forhold, B-bits kvantisering

- Signal til støy forhold: $SNR = P_S / P_N$:

$$SNR = 10 \log(P_S / P_N) = 10 \log(P_S) + 20 \log(L) - 20 \log(D) + 10.8$$

- Uttrykt ved antall bit, $20 \log(L) = 20 \log(2^B) = 6B$:

$$SNR = 10 \log(P_S / P_N) = 10 \log(P_S) + 6B - 20 \log(D) + 10.8$$

- $SNR \propto 6B$: 6 dB økning i SNR og dynamikk pr bit
- Hvis D er for stor \Rightarrow faller SNR, dvs hvis maks nivå sjelden overskrides faller SNR
- Avveining mellom mer kvantiseringsstøy eller flere feil pga metning av A/D-omformer

Kvantiserings effekter, eks 7.9a

- Nivå mellom $\pm 2V$, Hvor mange bit trengs for å få $< 5\text{mV}$ rms kvantiseringsfeil?
 - $D=4\text{ V}$ & $\text{rms} = \sigma = 5\text{ mV}$
 - $\sigma = \Delta/12^{0.5} \Rightarrow \Delta=12^{0.5} \cdot \sigma = 17.3\text{ mV}$
 - og $\Delta=D/2^B \Rightarrow 2^B = D/\Delta = 2000/17.3 = 231$
 - $B=\log_2(231) \approx 7.85$, dvs $B=8$ bit

Kvantisering av sinus

- $x(t) = A \cos(\cdot)$
- Effekt, $P_S = \text{effekt i signal} = A^2/2$
- Spiss-til-spissverdi: $D = 2A$

$$SNR = 10 \log(P_S) + 6B - 20 \log(D) + 10.8$$

$$SNR = 10 \log(A^2/2) + 6B - 20 \log(2A) + 10.8$$

$$SNR = 20 \log(A) - 3 + 6B - 20 \log(A) - 6 + 10.8 = 6B + 1.8$$

- Eksakt resultat: $SNR = 6B + 1.76 \text{ dB}$

7.5 Digital prosessering av analoge signaler

- Anti-aliasing filter, sampler (0-te ordens hold), kvantisering, koder:

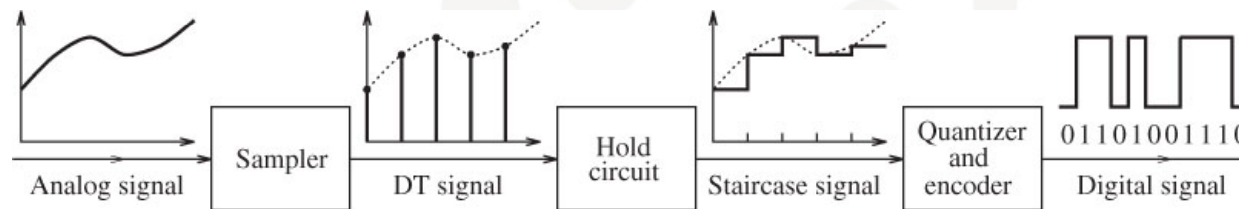


FIGURE 7.25 Block diagram of a system for analog-to-digital conversion. The sampler produces a discrete signal. The hold circuit yields the staircase approximation. The quantizer and encoder yield the digital signal as a stream of zeros and ones

- Så digital prosessering
- Dekoder, hold, analogt (anti-imaging) filter: $\sim \text{sinc}$:

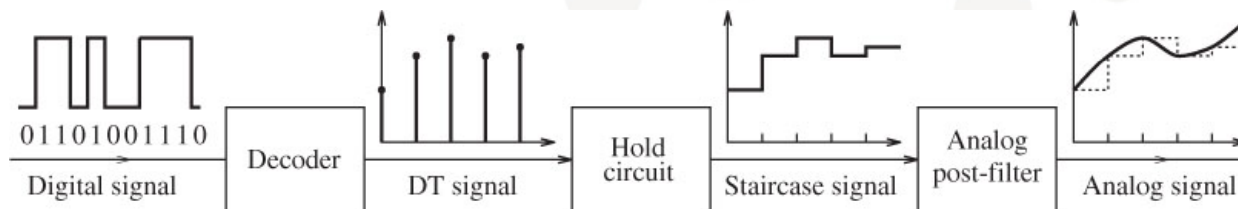
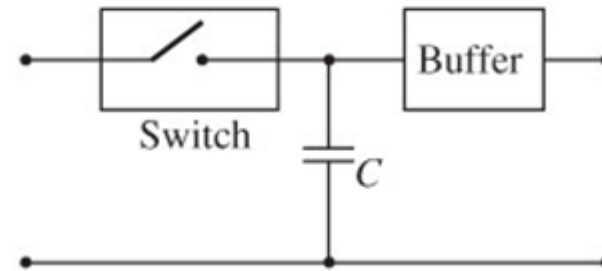


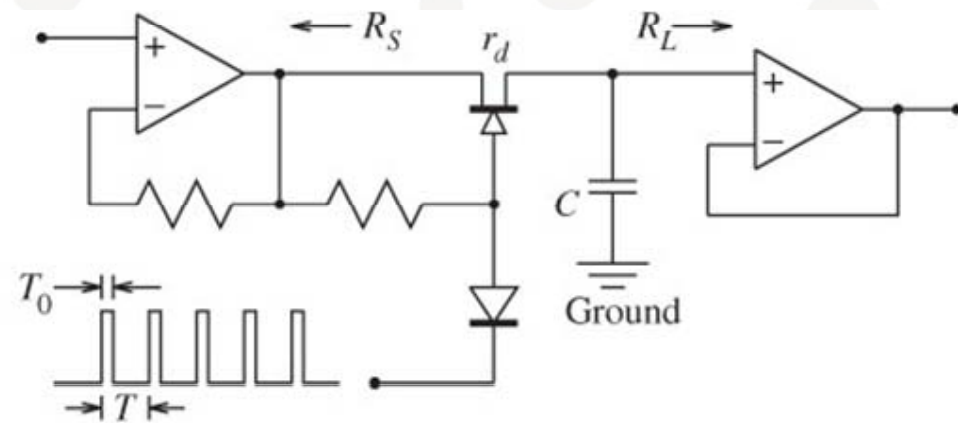
FIGURE 7.26 Block diagram of a system for analog-to-digital conversion. The decoder converts the digital bit stream to a discrete signal. The hold circuit yields a staircase approximation. The analog post-filter helps round out the edges to yield the smoothed analog signal

Praktisk A/D

- Klokke → Svitsj
 - Kondensator C lades hurtig
 - Utlades langsomt gjennom buffer-forsterker
- Parametre som påvirker ytelse:
 - Tid for å sample, $T_0 >$ en impuls
 - Tid for å bytte fra hold til sampling
 - Fall i Cs spenning i løpet av måletiden
 - Konverteringstid



Block diagram of sample-and-hold system

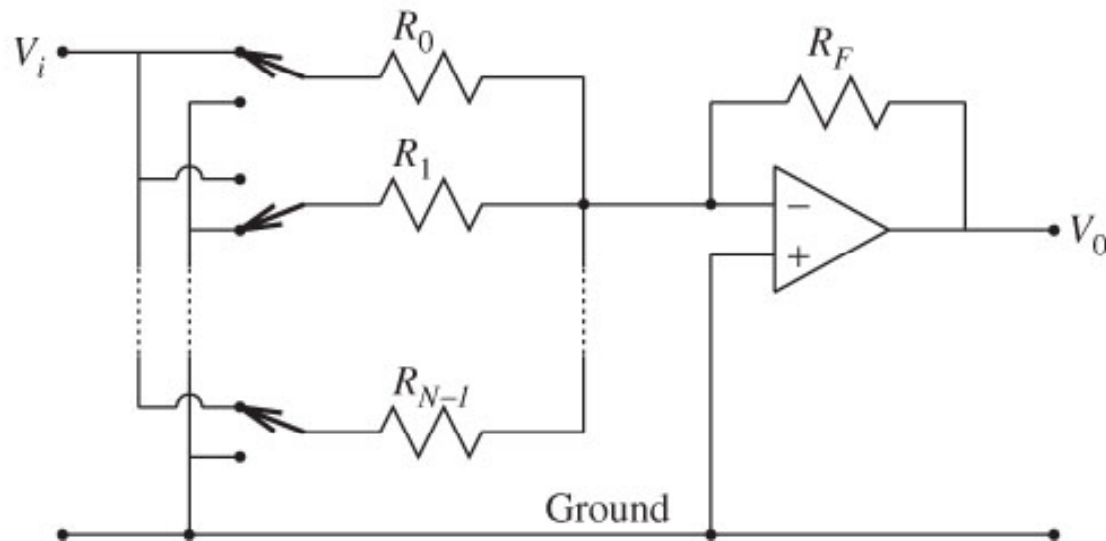


Implementation using an FET as a switch

Praktisk D/A

- Summasjonsforsterker: $V_O = -V_i (R_F/R_i)$
- R_i varieres ved å svitsje inn og ut motstander R_0 ... R_{N-1} avhengig av hvilket bit som er satt
- Linearitet avhengig av nøyaktighet på motstander

FIGURE 7.28 A
system for
digital-to-analog
conversion



Anti-aliasing filter

- Analogt filter
- Båndbegrensning av input signal: bare $f < S/2$
 - I praksis kan man ikke lage ideelle mursteinsfiltre
 - Blir derfor noe aliasing
- Mulig kriterium: energi over $S/2$ skal være mindre enn kvantiseringsstøyen
 - Kan avlede filterkrav av dette
- Kan og sette S litt høyere enn ideell verdi

Rekonstruksjonsfilter

- Analogt filter
- Ta ut sentrale delen av spektret, $|f| < S/2$
- Evt. korrigerer for 0-te ordens hold i D/A
- Oversampling (eks 4x)
 - Kan virke bortkastet
 - Gjør kravet til analogt filter mye lettere (Eks, neste slide)
 - Lenger avstand mellom repeterte spektra \leftrightarrow glattere trinn

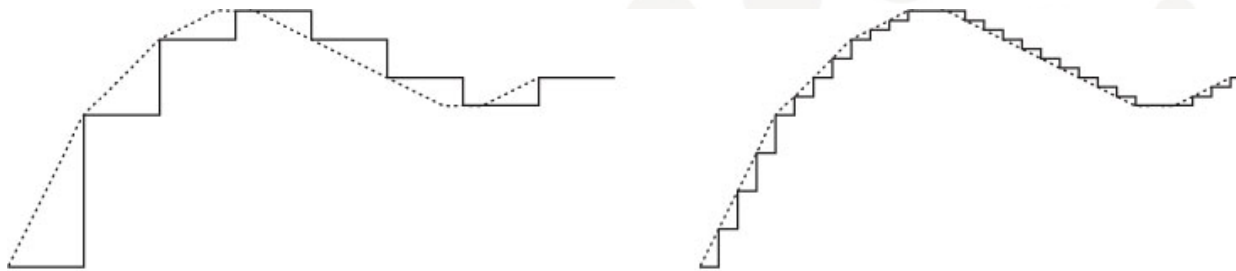
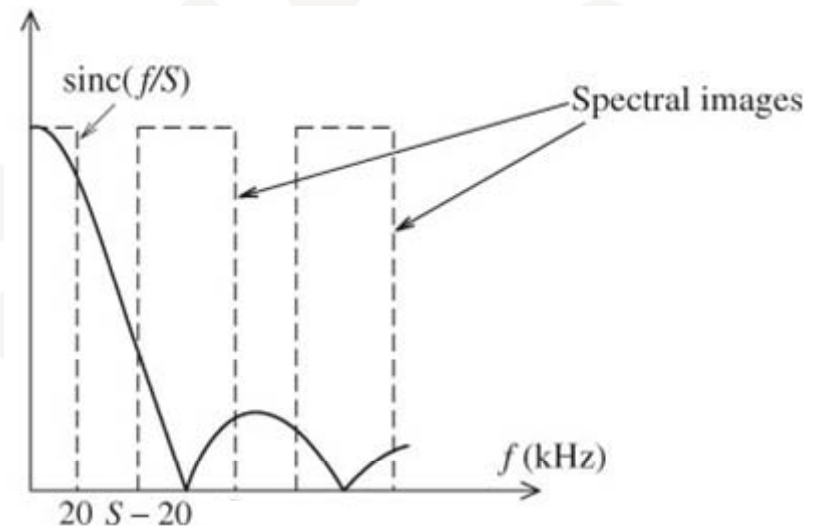


FIGURE 7.29 Staircase reconstruction of a signal at low (left) and high (right) sampling rates. Higher sampling rates produce a much better approximation to the underlying signal and allow the use of a simpler post-filter to smooth out the edges

Eks 7.12 Rekonstruksjonsfilter

- CD med $f_s=44.1$ kHz
 - Signal mellom 0 og 20 kHz
 - Maks passbåndsdemping: 0.5 dB
 - Stoppbåndsdemping >60 dB for $f > S-20$ kHz
- D/A, rett-fram sampling, $S=44.1$ kHz:
 - $S-20 = 44.1-20= 22.1$ kHz
 - Transisjonsbånd: 20-22.1 kHz \Rightarrow 80. ordens analogt Butterworth
- D/A, 4x oversampling, $S=176.4$ kHz:
 - $S-20=176.4-20 = 156.4$ kHz
 - Transisjonsbånd: 20-156.4 kHz \Rightarrow 4. ordens Butterworth
 - Enorm reduksjon i kompleksitet
 - Tidlige CD-spillere (4x + 2. eller 3. ordens Bessel \approx lineær fase))

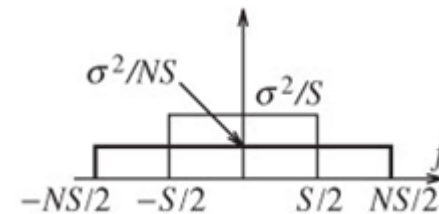


7.11 Multirate signalbehandling og Δ - Σ

- Fordel å redusere samplingsrate internt i DSP-er
 - Eks: ultralydmottaker for å få færre operasjoner
- Fordel å øke samplingsrate før rekonstruksjon
 - Eks: 4 x oversampling i CD-spiller
 - For å få enklere analog filter
 - Enklere kompensasjon for 0-te ordens hold

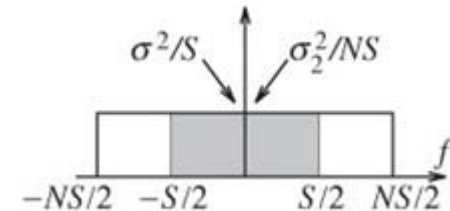
Kvantisering og oversampling

- Oversampling av analogt signal:
 - Enklere anti-aliasing filter
 - Færre bit i A/D-konverter da kvantiseringsstøy spres over et større frekvensbånd
- Kvantiseringsstøy ~hvit hvis nok bit brukes
 - Hvit \Leftrightarrow Jevnt spredt over alle frekvenser $-S/2 < f < S/2$
 - Varians $P = \sigma^2 = \Delta^2/12$
 - Spektraltetthet: $P_{ee}(f) = \sigma^2/S$
- Ved oversampling med N
 - Spektraltetthet: $P_{ee}(f) = \sigma^2/NS$



Oversamplingsfaktor vs antall bit

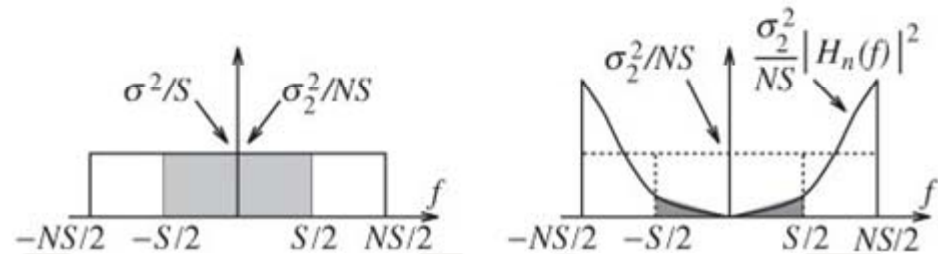
- Mål: kvantiseringsstøy i båndet $-S/2$... $S/2$ skal være det samme som før
- σ_2^2 skyldes bruk av $B_2 = B - \Delta B$ bit
- $\sigma^2/S = \sigma_2^2/NS \Leftrightarrow \sigma^2 = \sigma_2^2/N$
 - $\sigma^2 = D^2/(12 \cdot 2^{2B})$
 - $\sigma_2^2/N = D^2/(12 \cdot N \cdot 2^{2(B - \Delta B)})$
- Altså: $N = 2^{2\Delta B}$ eller $\Delta B = 0.5 \log_2 N$
 - $N=2$: dobling av rate \Rightarrow sparer halv bit
 - $N=4$: \Rightarrow sparer 1 bit
 - 1 bits CD, må spare $\Delta B = 15 = 0.5 \log_2 N$
 $\Rightarrow N = 2^{30} \sim 10^9$, $NS = 44.1$ teraHz!!!



Oversampling og støyforming

- Enda mer gevinst ved forming av støyspektret
- Vanlig støyformer gir sinus-form for $|f| < S/2$:

$$H_{NS}(f) = |2\sin(\pi f)/NS|^p$$

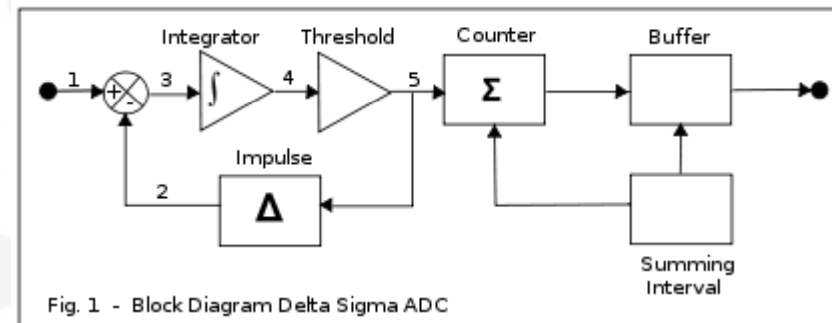


- p er orden til delta-sigma konverter
- Kan vise besparelse på $\Delta B = (p+0.5)\log_2 N - 0.5\log_2(\pi^{2p}/(2p+1))$ bit
- Eks: N=4, p=1: $\Delta B \approx 2$ bit
- Bedre enn ren oversampling

1-bits sampling (Sigma-delta) i CD

- CD: må ha besparelse på $\Delta B = 16 - 1 = 15$ bit
 - 1-bits A/D er bare en fortegnedetektor
- Hva skal til?
 - Oversampling: $N = 64$, dvs $44.1 \cdot 16 \text{ kHz} = 2.8224 \text{ MHz}$
 - Orden: $p = 3$. ordens støyforming
 - $\Delta B = (0.5 + 3) \cdot 6 - 3.55 = 17.5 \text{ bit} (> 15)$

- Sigma = integrator
- Delta = differanse



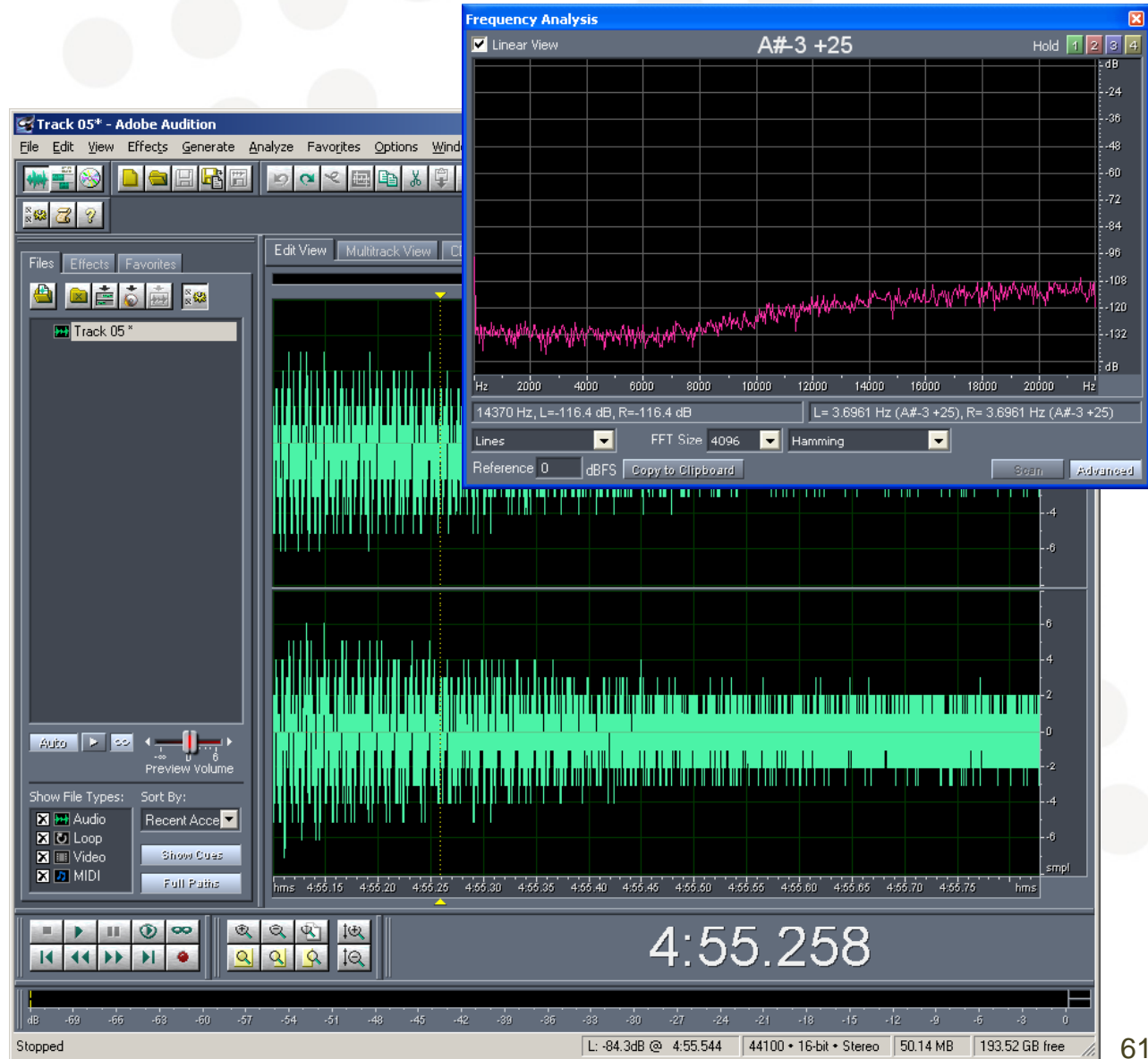
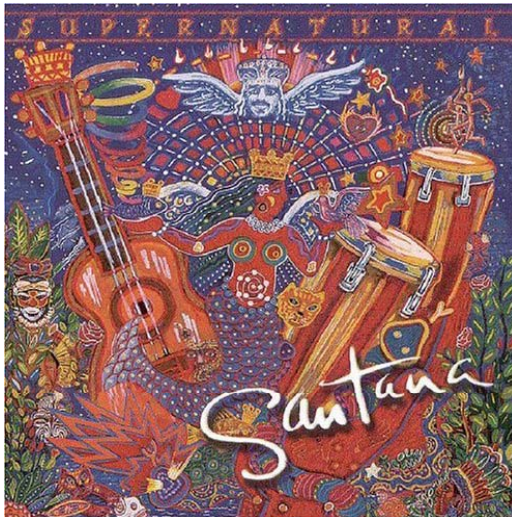
- III: Wikimedia commons

Dårlige CD-plater: Dither

- Tidlige CD-plater: avrundingsproblem.
- Også under prosessering og miksing med heltalls-aritmetikk
- Ikke som avrunding i kassa på Rimi
- => hørbar forringelse i de svake partiene.
 - Eks: pianotoner som dør ut
- Paradoks: må legge til kontrollert støy, dither, ved produksjon av CD-en.
 - Støyforming så dens energi er mest i diskanten

Dither

- Santana: Smooth, 1999
- Slutten av låten
- Maks +/- 3
- Trekant-fordeling
- Støyforming



Why 1-Bit Sigma-Delta Conversion is Unsuitable for High-Quality Applications

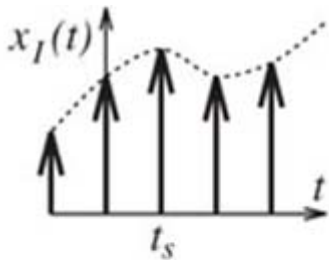
- S. P. Lipshitz & J. Vanderkooy, University of Waterloo, Ontario, Canada, Audio Engineering Society Convention, 2001
- Single-stage, 1-bit sigma-delta converters are in principle imperfectible. We prove this fact. The reason, simply stated, is that, when properly dithered, they are in constant overload. Prevention of overload allows only partial dithering to be performed. The consequence is that distortion, limit cycles, instability, and noise modulation can never be totally avoided.
- We demonstrate these effects, and using coherent averaging techniques, are able to display the consequent profusion of nonlinear artefacts which are usually hidden in the noise floor. Recording, editing, storage, or conversion systems using single-stage, 1-bit sigma-delta modulators, are thus inimical to audio of the highest quality.
- In contrast, multi-bit sigma-delta converters, which output linear PCM code, are in principle infinitely perfectible. (Here, multi-bit refers to at least two bits in the converter.) They can be properly dithered so as to guarantee the absence of all distortion, limit cycles, and noise modulation.
- The audio industry is misguided if it adopts 1-bit sigma-delta conversion as the basis for any high-quality processing, archiving, or distribution format to replace multi-bit, linear PCM.

22. oktober 2012

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Analog vs digital

- Hva vil dere svare til en slik påstand?
 - ”Digital lyd (CD) er klinisk og oppstykket – analog lyd (vinyl) er organisk”



Ideally sampled signal



Les vinyllydens revansj:

<http://blogg.uio.no/mn/ifi/innovasjonsteknologi/content/vinyllydens-revansj>