Response measurements

Overview and Practical Aspects

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Overview

Measurement methods

- Background
- Nondeterministic stimulus methods
  - 2 channel FFT
- Deterministic stimulus methods
  - impulse
  - noise/pseudorandom noise (FFT, MLS)
  - sinusoidal (stepped tone, multitone)
  - frequency sweeps (level recorder, TDS, Farina's method)

Aspects in response measurements

- Optimizing measurements
  - Averaging, system compensation, stimulus shaping
- The measurement environment
  - Effects and solutions

Response measurements

A large variety of alternative methods for measuring the *impulse response* or *frequency response* of a system.

$$x(t) \rightarrow \boxed{h(t)} \rightarrow y(t)$$

- the system is fed with an excitation signal $x(t)$
  - the signal must contain energy on all frequencies of interest

- the output $y(t)$ of the system is compared with the input using some method

Response measurements

Practical measurements always contain unidealities:

- **noise** - acoustical, electrical, thermal
- **nonlinearity** - distortion etc.
- **time variance** - temperature changes etc.
  -> the certainty of results is reduced

The stimulus should have high energy to achieve a sufficient signal-to-noise ratio (SNR) over the entire frequency range of interest.

Response measurements

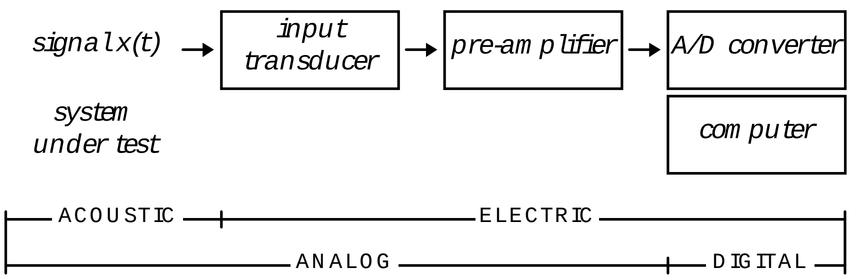
An overview of different response measurement methods is presented:

- Requirements and suitability
  - which method should I use?
- Properties
  - how does it work?
- Limitations
  - when are my results incorrect?

Demonstrations of methods and properties are shown along the way.

Background: Sampling

A typical modern signal analyzer:



Analog input signal
- length T [s]
- *bandwidth* B [Hz]

Sampling

- Sampling of a signal:
  - sample rate f_s
  - N samples taken
- no information is lost in the sampling process, if

$$N \geq 2BT$$

- the time between adjacent samples, ie. the signal time resolution is

$$T = N\Delta t$$

- the highest frequency that can be contained in the sampled signal is the *Nyquist frequency*

$$f_N = \frac{1}{2} f_s = \frac{1}{2\Delta t}$$

Sampling of a stationary signal

In practice, the length of a sampled block of data is limited in time:

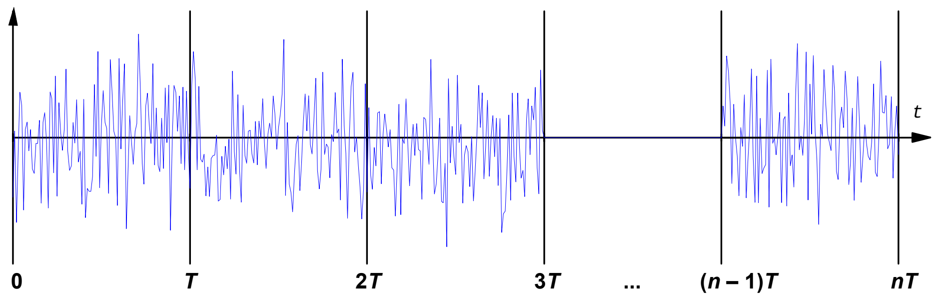
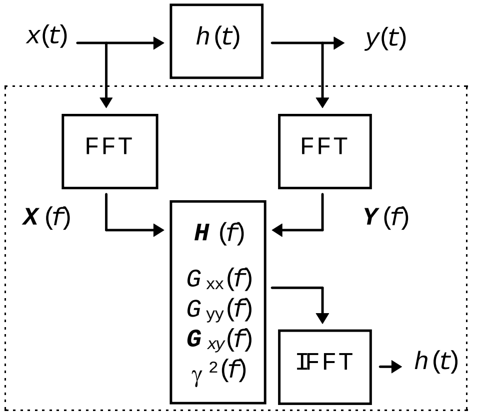


Figure. A time history is subdivided into n sample windows of length T.

FFT based measurements

2-channel FFT is a classic method for signal analysis

- suitable for systems where deterministic stimuli cannot be applied
- arbitrary stimulus, analysis between system input $x(t)$ and output $y(t)$



2 channel FFT: Properties

- Classic measurements performed with a 2-channel FFT analyzer
  - nowadays computers and soundcards are frequently used
- two channels required:
  - system input and output signals are analyzed simultaneously
- capable of measuring the complex frequency response
- nondeterministic stimulus
  - usually random noise (white or pink)
  - the spectral content of a limited time period varies and is unknown in advance
  - -> reasonable spectral accuracy requires time averaging
- arbitrary excitation signals can be used for measurement
  - applicable for systems where a stimulus signal cannot be fed

FFT analysis: Methods of estimation

- practical estimation of the signal autospectrum and cross spectrum:
  - expected value operation:
    1) Fourier transform of each data set (time window)
    2) averaging of the resulting raw spectra
- assuming stationary (and ergodic) random signals,
  - the analysis may be performed for data sets that are sampled adjacently in time one after another
- n data sets $x_k(t)$ of length T are collected
  - the total length of the analyzed signal is $T_{total} = nT$

FFT analysis: Methods of estimation

- the autospectrum for each signal $x(t)$ may be estimated as follows:

$$\hat{G}_{xx}(f) = \frac{2}{nT} \sum_{k=1}^{n} \left| \boldsymbol{X}_k(f,T) \right|^2$$

- the cross spectrum between two signals $x(t)$ and $y(t)$ may be estimated as follows:

$$\hat{\boldsymbol{G}}_{xy}(f) = \frac{2}{nT} \sum_{k=1}^{n} \boldsymbol{X}_k^*(f,T)\, \boldsymbol{Y}_k(f,T)$$

- the practical estimate for the coherence function is (by definition)

$$\hat{\gamma}_{xy}^2(f) = \frac{\left| \hat{\boldsymbol{G}}_{xy}(f) \right|^2}{\hat{G}_{xx}(f)\, \hat{G}_{yy}(f)}$$

FFT analysis: Methods of estimation

- If $n = 1$, a *totally meaningless result* is formed for all frequencies:

$$\hat{\gamma}^2_{xy}(f) \equiv 1$$

- The frequency resolution of the analysis is set by the individual data window length ($B = 1/T$), not the length of the complete data set $T_{total} = nT$.

FFT analysis: Transients and energy spectra

Power spectral density functions cannot be directly applied, if the analyzed signals are time limited, ie.

- $x(t) \neq 0$ and $y(t) \neq 0$ only when $0 \leq t \leq T$
- analysis would lead to a null average power density

In this case, spectral analysis is performed using signal energy densities instead of power densities.

- In practice, estimation is performed as before, but without the factors 1/T:

$$\hat{G}_{xx}(f) = \frac{2}{n}\sum_{k=1}^{n}\left|\boldsymbol{X}_k(f,T)\right|^2$$

$$\hat{\boldsymbol{G}}_{xy}(f) = \frac{2}{n}\sum_{k=1}^{n}\boldsymbol{X}^*_k(f,T)\,\boldsymbol{Y}_k(f,T)$$

2 channel FFT

Drawbacks and limitations:

- long averaging times required
  - unresponsive to abrupt changes in the system
- time windowing required to avoid spectral leakage
- SNR may vary between data blocks
  - noise-contaminated data blocks should be excluded from averaging to avoid errors in the frequency response
  - coherence function can be used to estimate response quality
- for acoustic measurements, the precise propagation delay of the acoustical transmission path must be known and the direct signal must be delayed to analyze the same parts of the excitation signal on both channels

2 channel FFT

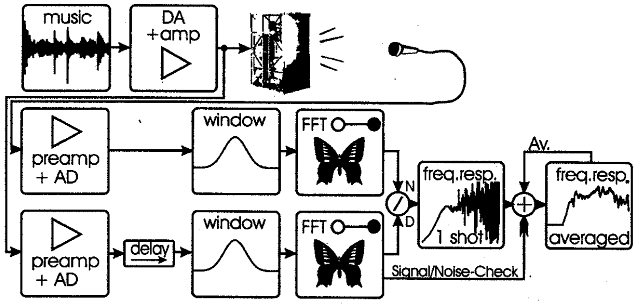


Fig. 2. Signal-processing steps for 2-channel FFT analysis with program material used as excitation signal.

Comparison with other methods:

- "old, slow and imprecise"…
- emphasis on complex frequency response, impulse response available with IFFT
- individual impulse responses cannot usually be time windowed prior to averaging (to remove unwanted reflections from response)

Windowing: Rectangular window

Rectangular window w(t):

$$w(t) = 1 \quad 0 \le t \le T$$
$$= 0 \quad \text{elsewhere}$$

- The continuous time data $v(t)$ is windowed to form the sampled data set $x(t)$:

$$x(t) = w(t)v(t)$$

- Time windowing has an effect on the resulting frequency response $\boldsymbol{X}(f)$:
  - multiplication in the time domain is seen as convolution in the frequency domain, ie.
  - the complex frequency response of the signal is convolved with th e Fourier transform of the time window:

$$X(f) = \int_{-\infty}^{\infty} W(\upsilon)V(f-\upsilon)\mathrm{d}\upsilon$$

Windowing: Rectangular window

The Fourier transform of a rectangular window is the *sinc function:*

$$W(f) = T\left(\frac{\sin \pi f T}{\pi f T}\right)e^{-j\pi f T}$$

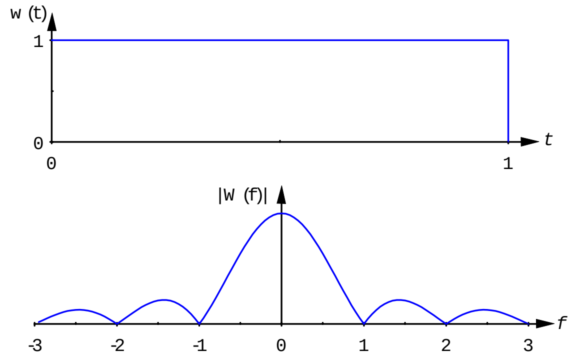


Figure. Rectangular window w(t) and the magnitude response of its spectral window |W(f)|. Time axis unit T (above), frequency axis unit 1/T (below).

Windowing

The magnitude component $|W(f)|$ forms the spectral window, which is of major importance in spectral analysis:

- strong sidelobes in $|W(f)|$ cause signal power to leak to adjacent frequencies far from the main lobe
- these can cause serious artifacts in spectral estimates especially for tonal or narrowband random signals
- a number of alternate windows are used to avoid the smearing caused by the rectangular window
  - used to force the signal amplitude towards zero at the edges of the time window
  - the data starts and stops smoothly, decreasing spectral leakage to the sides

Windowing: General issues

In practice, most FFT analyzers include at least three alternate windows:

- Rectangular window
  - for transients, which fit completely inside the window
- Hanning window
  - for random signals such as noise
- Gauss, flat top or other
  - intended for narrowband or sinusoidal signals

Windowing: Hanning window

- one of the oldest windows, still actively used today
- based on a cosine shape

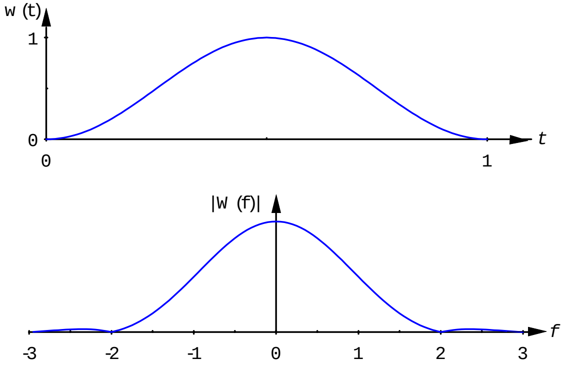


Figure. Hanning window $w_h(t)$ and the magnitude of its spectral window $|W_h(f)|$.

Windowing: Hanning window

- The Hanning window is defined as

$$w_h(t) = \frac{1}{2}\left(1 - \cos\frac{2\pi t}{T}\right) \qquad 0 \le t \le T$$
$$= 0 \qquad \text{otherwise}$$

- The Fourier transform of the Hanning window is

$$W_h(f) = -\tfrac{1}{4}W(f - f_1) + \tfrac{1}{2}W(f) - \tfrac{1}{4}W(f + f_1)$$

where $f_1 = 1/T$ and $W(f)$ is the Fourier transform of the rectangular window.

…

Figure. Examples of a sine signal and the spectral leakage of its signal power outside the actual frequency. Rectangular window, Hanning window. The original time data is a pure sinusoid, but it is truncated by the sampling window forming discontinuities at the edges.

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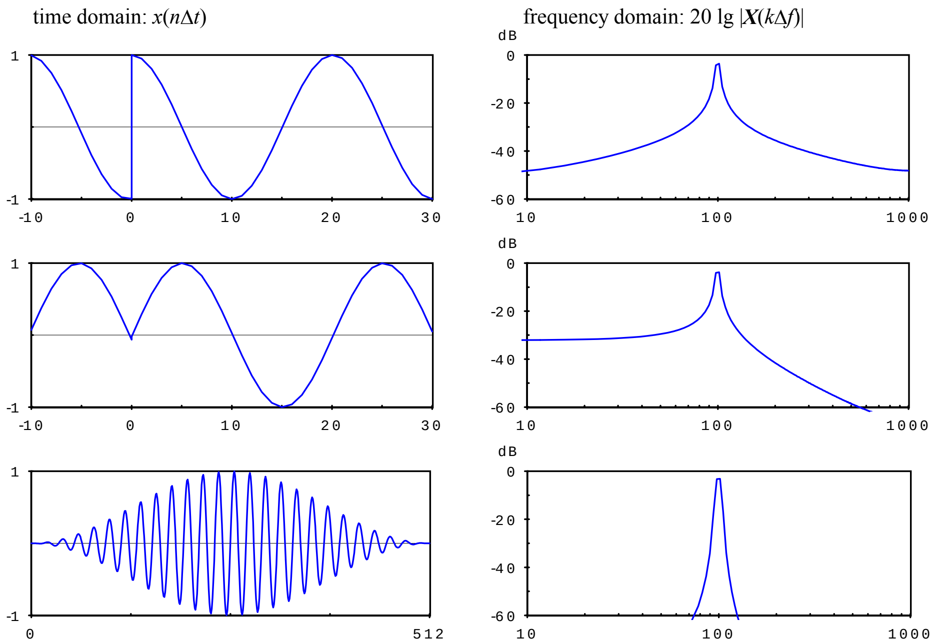


Gauss window

- based on the gauss curve
- no sidelobes in the frequency domain
- excellent amplitude accuracy
- the main lobe is wider than with the other windows
- suitable for (sparsely spaced) sinusoidal signals

Hamming window

- commonly used for speech analysis
- Hamming window + rectangular pedestal
- optimal resolution

Windowing: Comparison

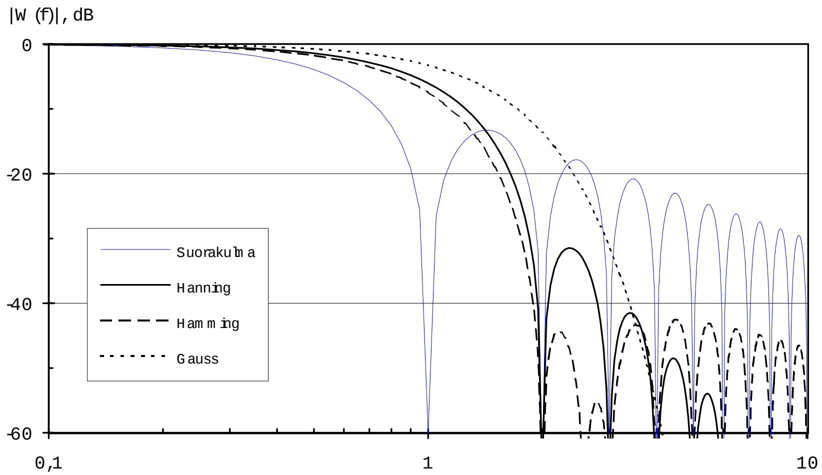


Figure. Comparison of the spectral shapes of the most common time windows. The frequency axis unit is $\Delta f = 1/T$.

Windowing: Comparison

A comparison of different time windows.

window	–3 dB bandwidth $\cdot\Delta f$	noise bandwidth $\cdot\Delta f$	1st sidelobe dB	sidelobes attenuate dB/decade	attenuation at frequency $0{,}5\cdot\Delta f$, dB
rectangular	0,9	1,0	–13	20	–3,9
Hanning	1,4	1,5	–32	60	–1,4
Hamming	1,3	1,4	–42	20	–1,8
Gauss	1,8	1,9	(ei)	(ei)	–0,9

Analysis methods related to FFT

- A number of common methods are available to improve the quality of the calculated spectral density and coherence estimates
  - zoom-FFT
    - (redundant with today's hardware)
  - overlap averaging
    - adjacent time windows are overlapped with each other (0…100%, typically 25-75%)
    - restores some of the statistic accuracy reduced by the smoothing window
  - frequency smoothing
    - applied in the post-processing stage after calculation of spectra, frequency response etc.
    - weighted averaging of n adjacent frequency components
    - commonly performed as convolution in the frequency domain

Frequency smoothing

A tradeoff between frequency resolution and statistical reliability of results:

- Example: smoothed autospectrum:

$$\langle G_{xx}(f)\rangle_f = \int_{-\infty}^{\infty} G_{xx}(\upsilon)W(f-\upsilon)\mathrm{d}\upsilon$$

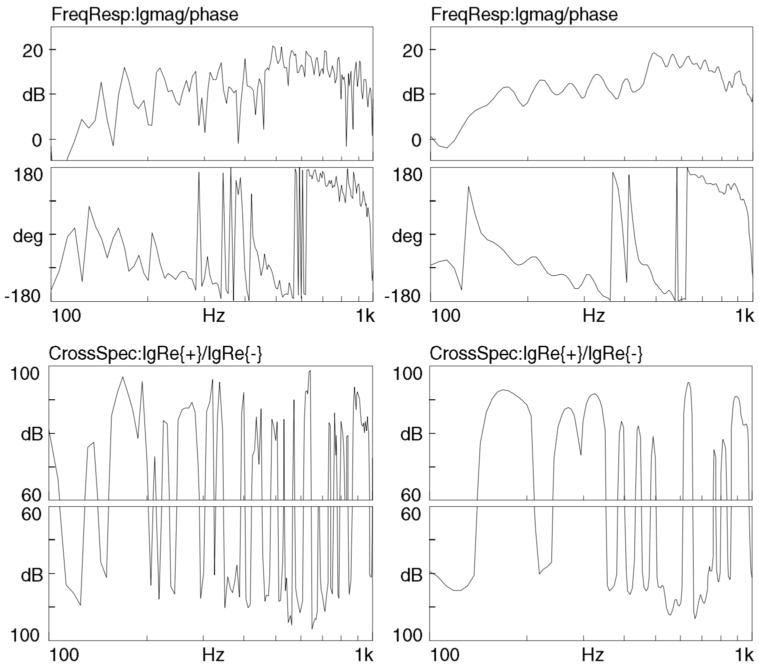
- a typical frequency smoothing window is the Hanning window (in the frequency domain):

$$\begin{aligned} W(\upsilon) &= \frac{1}{2}\left(1+\cos\frac{2\pi\upsilon}{B}\right) \qquad -\frac{B}{2}\le \upsilon \le \frac{B}{2} \\ &= 0 \qquad \text{elsewhere} \end{aligned}$$

where B is the bandwidth of the smoothing window

…

Figure. Two examples of frequency smoothing: Complex frequency response and an intensity spectrum (calculated from the cross spectrum). Smoothing applied using a Hanning window with a bandwidth of 7 frequency components.



Random errors

Applicable for FFT methods and random data.

- inaccuracies caused by random errors in the data are require spectral analysis to be performed as an average over *n* pieces of raw data
- formulas for normalized random error ε_r.
  - ie. the distribution of the random error divided by the measurement result
  - inaccuracy given as $\hat{x} \pm \varepsilon_r \cdot \hat{x}$, where $\hat{x}$ is the measurement result (approximating the actual value)
  - for the autospectrum $\hat{G}_{xx}(f)$:

$$\varepsilon_r = \frac{1}{\sqrt{n}}$$

  - for the cross spectrum magnitude $\left|\hat{G}_{xy}(f)\right|$:

$$\varepsilon_r = \frac{1}{\left|\hat{\gamma}_{xy}(f)\right|\sqrt{n}}$$

Random errors

- for the coherence function $\hat{\gamma}^2_{xy}(f)$:

$$\varepsilon_r = \frac{\sqrt{2}\left[1 - \hat{\gamma}^2_{xy}(f)\right]}{\left|\hat{\gamma}_{xy}(f)\right|\sqrt{n}}$$

The equations above clearly show the importance of averaging:

- effect of the number *n* of raw sampled data sets
- if power density estimates are formed without averaging, the normalized random error becomes one, ie. the normalized estimate yields the value 1 ± 1 (!)
- potential coherence problems are not seen without averaging
- normalized inaccuracies for autospectral estimates:
  - $n = 1$: 1 ± 100 %,
  - $n = 10$: 1 ± 32 %,
  - $n = 100$: 1 ± 10 %,

Deterministic methods

Most methods for system response measurement use a *deterministic stimulus*:

Impulses
- Impulse method

Random/pseudorandom signals
- MLS
- Single channel FFT

Sinusoidal stimulus
- Stepped sine
- Multitone
- Frequency sweeps
  - Traditional level recorder
  - TDS
  - Log sweep method

The impulse method

- an impulse response is naturally acquired by exciting a system with an impulse
  - no additional processing is required
- the frequency response available directly by FFT
- the stimulus is ideally a Dirac delta function
  - zero length, infinite amplitude
  - -> in practice, a band and amplitude limited approximation is used
- impulse excitation yields SNR problems for most systems
  - the stimulus has a very high crest factor and very little energy

The impulse method

- impulse trains
  - synchronous averaging increases SNR
  - results in periodic IR
  - danger of time aliasing: the impulse response must completely fit in the interval between two impulses
- unidealities in the excitation signal may be compensated for:
  - reference response is measured by linking the system input and output
  - measured responses are compensated with reference response (by deconvolution, ie. division in frequency domain)

The impulse method

- properties
  - a very fast and simple method
  - rather immune to time variance
- limitations
  - low noise floor required
  - distortion components cannot be separated or removed from responses
- applications
  - suitable for purely electrical measurements (no acoustic path)
  - also suitable for loudspeaker testing (under quiet conditions): very high voltages can be applied to element, as voice coils do not have the time to overheat (the impulses have a limited energy)

The MLS method

- stimulus based on Maximum Length Sequences (MLS)
- method originally proposed by Manfred Schroeder in 1979
- measures (periodic) impulse response,
  frequency response available using FFT
  - MLS stimulus
    - binary sequences of length $L = 2^n - 1$
    - simple to create using a binary shift register with n bins
    - sequence exhibits close to unity autocorrelation,
      ie. a flat white magnitude spectrum (but no DC component)
    - pseudorandom phase spectrum
    - sequence contains 2^n times more signal energy than a single impulse, as the excitation is stretched out over the whole measurement period

The MLS method

MLS deconvolution

- the IR is recovered using periodic cross correlation
  - the impulse response is also periodic
  - time-aliasing error results, if the system response is longer than sequence length L
- may be rapidly computed using the Fast Hadamard Transform (FHT)
  - calculation entirely in the time domain
  - computation involves only addition and subtraction; no multiplications required!
- originally MLS was an elegant method with the limited hardware and computing power of the 1980's
- the deconvolution yields a periodic impulse response:

The MLS method

Practical considerations:

- a pure pulse train signal would be an inpractical stimulus
  - pulses contain very little energy
  - signal exhibits an exceedingly high bandwidth

- in a hardware MLS generator, the signal level is kept constant between two pulses using a first-order hold circuit
  - theoretically ideal crest factor of 0 dB
  - in practice, any following filtering of the signal will increase the crest factor
  - time windowing results in a sinc(x) aperture loss in the output spectrum: -4 dB at $f_s/2$ (applicable frequency range is limited to $f_s/3$)

The MLS method

  - in an oversampling audio DA converter, the spectrum remains flat
    - however, the converter exhibits a (symmetric) sinc-shaped impulse response in the time domain
    - the oversampling stage anti-aliasing filter may induce a large overshoot (5-8 dB) of the MLS output, ie. the output cannot be fed at full level without a risk of distortion

- preemphasis
  - the MLS spectrum can be colored with an appropriate emphasis
  - stimulus preemphasis should be removed from measured responses (by deconvolution, ie. division in frequency domain)

The MLS method

Advantages

- effective noise cancellation
  - uncorrelated energy (noise, disturbances) is phase randomized uniformly along the resulting IR time axis
  - postaveraging may be used to yet reduce the IR noise floor
- compared to a single pulse, the MLS exhibits a theoretical SNR improvement of 20 log(2^n) dB, ie. +96 dB for n = 16!

Disadvantages

- high vulnerability to distortion and time variance
- inherently white excitation spectrum
- distortion artifacts produce false peaks in the response
- for optimum IR peak-to-noise ratio, the stimulus level must be carefully adjusted by trial-and-error for each measurement situation

The MLS method

Conclusions

- The MLS method is theoretically very promising, but is limited in practice due to unidealities in real systems
  - in practice, the MLS measurements are very sensitive to distortion and time variance
  - the dynamic range is practically limited to 60-70 dB regardless of the measurement system properties
  - noise rejection properties are generally very good
  - systems with fair amounts of distortion can be measured, but the resulting impulse response is far from optimal
  - theoretical maximum SNR is not realistically attainable, as small amounts of distortion and/or time variance are present in almost any analog or acoustical system

Periodic signals of length 2^n

- resembles 2 channel FFT but with some exceptions:
  - deterministic stimulus (which is fed to system)
    - arbitrary stimulus signal: noise, sweep, music etc.
    - sufficient signal energy required over bandwidth of interest
    - stimulus length 2^n samples (suitable for FFT analysis)
  - only a single channel is required
  - far better precision than with undeterministic stimuli

- frequency response is acquired by division: $\boldsymbol{H}(f) = \boldsymbol{Y}(f) / \boldsymbol{X}(f)$
  - nonwhite stimulus magnitude and phase are compensated for automatically

- measurement can be enhanced by appropriate choice of stimulus signal
  - synthezised random noise (arbitrary magnitude, random phase)
    - FFT synthesis allows flexible manipulation of stimulus spectrum
    - measurement requires at least two periods of repetition
      (FFT assumes a repeated signal to yield the required magnitude spectrum)

The stepped sine method

- pure sine tones used for excitation
- transfer function measured step by step at individual frequencies

- very slow
  - in practice, the spectral resolution is limited by the total time available for measurements
  - the system must each time be enabled to settle to steady state prior to proceeding with a measurement of the next frequency (very slow with resonant high-Q systems)

- very accurate:
  - completely suppresses surrounding frequencies, excellent SNR
  - stimulus has low crest factor (3 dB)
  - measurement certainty and repeatability can be very high

    …

The stepped sine method

…

- if the system output is analyzed using FFT, the sine must be exactly periodic with the FFT block length to avoid spectral leakage due to the rectangular time window

-> in practice, this requires synchronized digital generation of the sine stimulus

- disadvantages
  - for acoustic measurements, reflections can only be gated out if the analysis time window is shorter than the reflections' relative propagation time delay

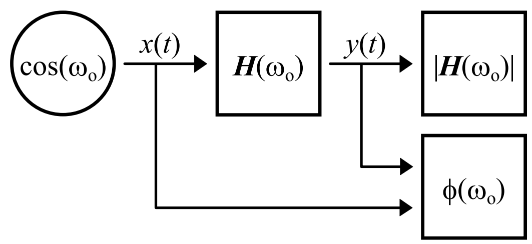
Multitone methods

- extensions of the stepped sine method
- two or more sinusoids using different frequencies and relative phases are applied simultaneously
- analysis using FFT

- faster response measurements
- enhanced distortion measurements
  - intermodulation distortion with two sines
- arbitrary spectra can be created using a large group of closely spaced tones
  - simulation of for instance a speech spectrum for testing of telecommunications systems
  - background noise levels may be monitored between tones (suitable for measuring compressor/expander noise performance)

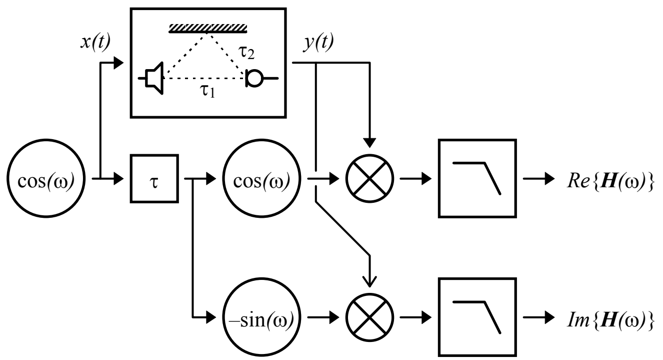
Traditional frequency sweeps: The level recorder

- a traditional method for recording the frequency response magnitude
- suited for analog technology: sweep generator connected to level recorder (pen plotter)
- logarithmic sweep
- noise or reflections cannot be suppressed
- simple flattening of frequency response by mechanical limits of pen velocity
- phase information is unavailable without a synchronized phase meter



Time Domain Spectrometry (TDS)

- another method for measuring transfer functions with frequency sweeps
- capable of measuring the full complex transfer function
- originally devised by Richard Heyser for the measurement of loudspeakers



Time Domain Spectrometry (TDS)

- main components:
  - sweep signal generator (producing phase locked sine/cosine signals)
  - two multipliers (to produce the sum and difference of input signals)
  - two low pass filters (to pick the difference signal from the multiplier output)
  - two time delays (to account for propagation delays)
  - -> suitable for analog technology

Time Domain Spectrometry (TDS)

- measurement of Re{$\boldsymbol{H}(f)$}:
  - system is excited with sine stimulus from generator
  - system output is multiplied with sine stimulus
  - the multiplier output is lowpass filtered with a fixed cutoff frequency
- measurement of Im{$\boldsymbol{H}(f)$}:
  - system is excited with sine stimulus from generator
  - system output is multiplied with the 90' phase shifted cosine stimulus
  - the multiplier output is lowpass filtered with a fixed cutoff frequency
- generator signals can be delayed prior to account for a certain propagation delay in the system
  - analog means for time windowing
  - the direct sound response or an individual reflection's response can be extracted using the delay

Time Domain Spectrometry (TDS)

- properties
  - tolerant to distortion and noise outside the instantaneous sweep frequency
    - artifacts are attenuated by the LPF's
  - simulated quasi-free-field measurements are possible with correct sweep rate and LPF cutoff frequency
- limitations:
  - linear sweeps, white spectrum
    - SNR problems at low frequencies
    - example: 1 s sweep 20-20kHz -> only 4 ms energy for 20-100 Hz frequency range!
    - requires very long sweep or repeating the measurement for multiple frequency ranges

Time Domain Spectrometry (TDS)

- LPF filters cause limitations
  - LPF responses limit resolution and accuracy of the measured frequency response components
  - ripple is not properly removed from responses at low frequencies
  - -> can be solved using a double-excitation "mirrored" measurement scenario, where the LPF's may be omitted
  - slower measurements, intermediate storage of results is required
- ripple near sweep start and end frequencies due to abrupt start and stop of signal
- unintuitive complexity: the relations between TDS sweep rate, LPF cutoff frequency and achieved results are not immediately evident but require calculations

Time Domain Spectrometry (TDS)

- comparison to other methods
  - double-excitation comparable to MLS or FFT methods using periodic stimuli, where two cycles of stimulus must be used
  - sinusoidal stimulus has low crest factor (3 dB), compared to MLS (in practice >8 dB)

Nonperiodic sweeps

- spectral synthesis:
  - arbitrary magnitude + random phases yield a noise signal in the time-domain
  - an increasing group delay in the phase spectrum results in a sweep in the time-domain
- with sweeps, repetition is not required for a certain spectral content
  - stimulus signal can be emitted only once, and the response analyzed immediately
- harmonic distortion components can be isolated entirely from the acquired IR
  - results in a purely linear IR regardless of system distortion
  - requirement for long sweeps (> IR length) and a large FFT size
- acquisition must be continued after the sweep end to capture all delayed components

Nonperiodic sweeps

- log sweeps are more robust than MLS
  - slow sweeps are quite insensitive to time variance
  - even heavily distorted systems can be measured accurately
  - capable of simultaneous measurement of the linear transfer function and frequency-dependent distortion components
  - noise rejection is only fair
  - reverberant measurement conditions require rather long sweeps

Log sweep using Farina's method

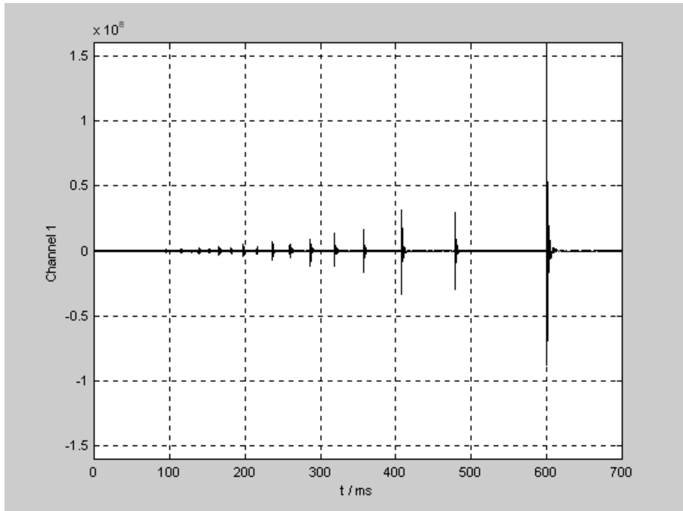
- a novel method: first published on in 2000 by Angelo Farina
  - several AES journal articles have been published since

- based on a nonperiodic logarithmic sweep stimulus
  - sweep spectrum can be synthesized to arbitrary requirements
    - sweep energy can be made independent of signal amplitude
  - an inverse filter is calculated from the stimulus signal
  - system output is convolved with the inverse filter

- method outcome:
  - a time-delayed impulse response,
  - preceeded by responses of harmonic distortion orders, separated in time(!)

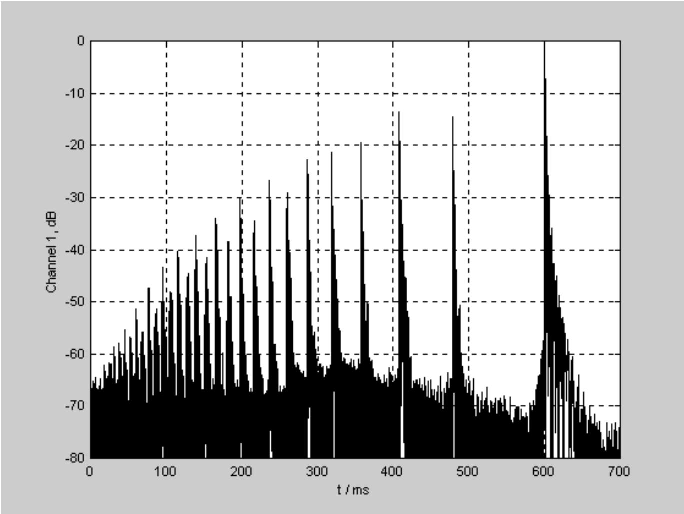
Log sweep using Farina's method

Impulse response: distorting tube guitar amplifier, anechoic conditions



Log sweep using Farina's method

Energy-time curve: distorting tube guitar amplifier, anechoic conditions



Log sweep using Farina's method

- properties
  - harmonic distortion can be accurately measured together with linear response
  - linear response and each harmonic distortion order are separated in time
  - linear response can be extracted from distorted measurement
  - distortion orders can be analyzed separately to provide narrowband magnitude spectra for each order

- time aliasing between adjacent responses must be avoided
  - number and length of required distortion orders in response must be accounted for when choosing sweep length
  - reverberation, resonance etc. is seen in each adjacent response
  - response tails should not overlap with adjacent responses
  - practical choice again related to background noise levels in response

Aspects in response measurements

Optimizing measurements
- Averaging
- System compensation
- Nonlinearities
- Time variance

Effects of the measurement environment
- Noise
- Reflections
- Room modes
- …solutions and workarounds

Optimization: Averaging

Synchrononus averaging can increase measurement SNR:
- the measurement is repeated and the results are averaged together
- theoretically, the SNR increases 3 dB for each doubling of averages

(the signal is correlated, but random noise isn't)
- stationary measurement conditions are required

Nonlinear or time variant systems
- averaging will not increase SNR
- this is a good method to check if a system non-LTI

Preaveraging
- system output is averaged prior to deconvolution
- less processing required (deconvolution required only at end of measurement)
- suitable for stationary conditions

Optimization: Averaging

Postaveraging

- separate deconvolved responses are averaged together
- suitable for MLS measurements, as the deconvolution spreads noise and disturbances randomly in time
- may be used for analyzing slowly varying nonstationary conditions
  - averaging is performed over a long period of time
  - system is assumed adequately stationary during a single response measurement
  - individual responses are compared prior to inclusion into averaging
    - if differences are too large, corrupted responses may be rejected from average

Optimization: System compensation

The response of the measurement system can be compensated for:

1) measurement of reference response
   - electrical loopback (output to input)
     - response of analog components and converters
   - acoustical compensation (source to microphone)
     - response of transducers
   - environmental compensation
     - response of floor reflection etc.

2) storage of reference response
   - any later changes in the system must be accounted for!

3) compensation of measured responses with reference response
   - deconvolution (FFT, response/reference, IFFT)

Optimization: Equalization

Pre-equalization

- stimulus is equalized to suit requirements:
  - transducer magnitude response (flat, speech spectrum etc.)
  - spectral shaping of stimulus to account for background noise
- measured responses must be compensated for using inverse filtering
- response SNR may be greatly improved
  - for room measurements, pink or red MLS works better than white
  - source distortion should be considered when increasing gain!
    - no amount of eq can change the loudspeaker sensitivity
    - if the source distorts heavily, MLS won't work, but logsweep will

Post-equalization

- responses can naturally be equalized also after measurement
  - flattening of loudspeaker response for auralization purposes etc.
- response SNR cannot be improved at this stage

Optimization: Novel methods

- For sweep stimuli, the spectral content can be adjusted using
  - amplitude shaping (adjust gain at frequency band *b*)
  - time stretching (adjust sweep rate at frequency band *b*)

- Time stretching
  - variable sweep rate enables arbitrary spectral content using unity (or arbitrary) gain at any frequency band
  - stimulus signal may become excessively long…

Effects of nonlinearities

Most IR measurement methods assume a LTI system!

- common causes for nonlinearities
  - transducer distortion (mainly loudspeakers)
  - signal dynamics processing (compressors, expanders, limiters, gates)
  - signal clipping (digital/analog)
  - distortion in amplifiers etc.

- results of a distorting system vary between methods
  - FFT: coherence decreases
  - MLS: spurious peaks in response
  - logsweep: distortion orders show up as separate responses

- detecting distortion
  - response noise level increases with stimulus level increase
  - averaging does not improve response peak-to-noise level

Effects of time variance

Time variance
- again, most IR measurement methods assume an LTI system
- common causes for time variance
  - loudspeaker coil heating (magnetic properties vary with temperature)
  - moving people or objects in a room
  - (smaller effects: changes in enviromental temperature and humidity)
  - air flow (heavy ventilation, wind)
  - cheap computer sound cards!
    - sample/sync dropouts cause ridiculous results with MLS
  - compressors

Effects of the acoustical environment

Noise
- can be variable with time, frequency and measurement position
- stationary, impulsive or combination

Reflecting surfaces
- effects on frequency response
  - time lag causes comb filtering
- effects on impulse response
  - reflections limit the analysis of direct sound

Sound fields
- free field
  - frequency limits (low&high), bounded area
- diffuse field
  - frequency limits (low), bounded area
  - local variations possible
  - reverberation time is only defined for a diffuse field…

Effects of the acoustical environment

Room modes
- cause steep frequency dependent deviations in sound pressure level in different room locations
- pressure maxima near boundaries and especially at corners
- spatial averaging is required for an adequate estimation of diffuse field conditions
- mode density increases with frequency
  - few modes spaced widely apart at low frequencies, cause large deviations
- local peaks and knots affect frequency response
- beating can occur due to interaction (exchange of energy) between closely-spaced modes
- room modes do exist regardless of the room shape!

Working with the acoustical environment

Noise

- consider airborne and structureborne noise separately
- suppress noise sources
- isolate measurement environment
- choose noise tolerant measurement methods
  - adjust stimulus spectral content
  - take more averages

Working with the acoustical environment

Reflections

- check reflection paths for each surface (between source and receiver)
  - is there a reflection path from source to surface to receiver?
  - estimate reflection path length, relative propagation time delay and resulting notch frequency
- estimate size and shape of reflecting surface
  - flat/concave/convex?
  - single small surfaces do not reflect much at low frequencies
  - estimate reflective frequency band from surface dimensions (surface reflects, when d > lambda/10)

Working with the acoustical environment

- optimize measurement environment geometry
  - remove reflecting surfaces if possible
  - move surfaces/source/receiver to more favorable positions
- cover remaining reflecting surfaces with absorbing material
  - required effective absorbent thickness depends on surface dimensions and measurement frequencies
- take reflections into account in the analysis stage
  - removal by time windowing
    - limited frequency resolution
  - removal by deconvolution
    - requires stationary measurement setup

Working with the acoustical environment

Free field conditions

- check validity of freefield conditions
  - point source (tweeter etc.)
  - measure sound pressure level at doubling distances
  - does the level decrease by 6 dB/doubling of distance?
- different directions may yield different results
- boundary limits of measurement area

Working with the acoustical environment

Room modes

- adjust measurement setup geometry
  - move source/receiver to find a flatter response

- increase absorption/diffusion of reflecting surfaces
  - increased absorption can result in more distinguished room modes due to the remaining reflective surfaces!
  - take frequency-dependent properties into account when choosing materials and structures
    - thin materials may only increase the difference between low and high frequencies

Conclusions

- Choose an appropriate measurement method
  - deterministic/nondeterministic stimulus?
  - requirements: speed, accuracy, tolerance, distortion measurement?

- Investigate the system under test and the measurement environment
  - note potential problems or limitations
  - distortion? time variance?
  - reflections?

- Select and test measurement settings accordingly
  - use averaging, emphasis etc. if possible

- Go forth and measure…

— *Thank you!*