PA systems

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Acoustics II:
public address systems

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introduction
introduction

- aim:
  - generation of sufficiently high sound pressure ($S/N > 10\ldots25\ \text{dB}$) in the audience area
  - homogeneous level distribution ($\pm 3\ \text{dB}$)
  - good direct sound supply
  - suitable audio quality

- means:
  - loudspeaker
introduction

▶ applications:
  ▶ source is too weak
  ▶ source is too omnidirectional (strong excitation of diffuse field in a room)
  ▶ non-acoustical source
PA systems for speech
PA systems for speech:

- auditoria
- churches
- public buildings (e.g. emergency announcements ...)

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PA systems for speech: speech signals

- sound pressure level of speech in 1m distance: → 65 dB(A)
- spectrum:

![Chart](image)
relevance of third-octave bands for intelligibility: [Bar chart showing the relevance of different frequency bands for intelligibility.]
PA systems for speech: necessary frequency range

- suitably adjusted system frequency response for speech:
PA systems for speech: elements

- microphone
- amplifier
- loudspeaker
- room
PA systems for speech: feed-back

- feed-back loudspeaker → microphone:
  - amplitude response distortions
  - temporal stretching of transient signals
  - possible instability (whistling sound)

- target: about 10 dB margin to the point of instability
maximal amplification
maximal amplification: system analysis

$n$ identical loudspeakers

- microphone gets direct sound from $L_1$ and diffuse sound from $L_2 \ldots L_n$
- listener gets direct sound from $L_1$ and diffuse sound from $L_2 \ldots L_n$
maximal amplification: system analysis

idealizations:

▶ direct sound:
  ◀ point source behavior $\rightarrow p \sim \frac{1}{r} \rightarrow -6$ dB per doubling of distance

▶ diffuse sound:
  ◀ constant in the room
maximal amplification: block diagram

G3 transfer speaker $\rightarrow$ microphone

G2 transfer microphone signal $\rightarrow$ listener (by loudspeaker)

G1 feed-back microphone signal $\rightarrow$ microphone signal (by loudspeaker) $\leq 0.1$
maximal amplification

maximal amplification \( G_{\text{MAX}} = \frac{p_{E, \text{withLS}}^2}{p_{E, \text{withoutLS}}^2} \)

with \( G1 = 0.1 \) follows for \( G_{\text{MAX}} \)

\[
G_{\text{MAX}} = 1 + 0.1 \frac{1}{d_{QM}^2} \left( \frac{1}{d_{LSE}^2} + n \frac{16\pi}{AQ} \right) \left( RW_{LS}(\gamma) RW_{M}(\beta) \frac{1}{d_{LSM}^2} + n \frac{16\pi}{AQ} \right)
\]

- \( d_{QM} \): distance source - microphone
- \( d_{LSE} \): distance loudspeaker - listener
- \( d_{QE} \): distance source - listener
- \( d_{LSM} \): distance loudspeaker - microphone
- \( n \): number of active loudspeakers
- \( A \): total absorption
- \( Q \): loudspeaker directivity (re. full solid angle)
- \( RW_{LS}(\gamma) \): loudspeaker directivity (attenuation in direction \( \gamma \))
- \( RW_{M}(\beta) \): microphone directivity (attenuation in direction \( \beta \))
maximal amplification

▶ for applications of interest holds:
  ▶ \( G_{\text{MAX}} \gg 1 \)
    ▶ reasonable usage
  ▶ \( \frac{1}{d_{\text{LSE}}^2} \gg n \frac{16\pi}{AQ} \)
    ▶ distance speaker - listener \(<\) critical distance with consideration of \( Q \)
  ▶ \( n \frac{16\pi}{AQ} \gg RW_{\text{LS}}(\gamma)RW_M(\beta)\frac{1}{d_{\text{LSM}}^2} \)
    ▶ feed-back dominated by diffuse field
maximal amplification

- from the above follows approximately:
  - \[ G_{\text{MAX}} \sim \frac{1}{d_{QM}^2} \]
  - maximal amplification is inversely proportional to the square of the distance source - microphone
  - \[ G_{\text{MAX}} \sim \frac{1}{n} \]
  - maximal amplification is inversely proportional to the number of loudspeakers
  - \[ G_{\text{MAX}} \sim Q \]
  - maximal amplification is proportional to the loudspeaker directivity
maximal amplification: example

- auditorium:
  - $V$: 2000 m$^3$
  - $RT$: 2 s $\rightarrow$ $A$: 160 m$^2$

- geometry of the system:
  - $d_{QM}$: 0.3 m
  - $d_{QE} = d_{LSE} = d_{LSM}$: 15 m

- loudspeaker directivity:
  - $RW_{LS}(\gamma)$: 1
  - $Q$: 1

- microphone directivity:
  - $RW_{M}(\beta)$: 1

- maximal amplification $= 6$ dB
maximal amplification: example

- maximal amplification = 6 dB is not sufficient
- need for more directional microphones and loudspeakers
focusing loudspeakers
horn speakers
horn speakers

▶ properties:
  ▶ increased efficiency due to better impedance matching
  ▶ controlled directivity
horn speakers

- example mid-range horn, 70x40x40cm:
column speakers
**column speakers**

- arrangement of several chassis in a row
- focusing in direction of the row

Example: column of length 1.3 m with 9 chassis:

<table>
<thead>
<tr>
<th>Frequency (Hz)</th>
<th>Q</th>
<th>250 Hz</th>
<th>500 Hz</th>
<th>1 kHz</th>
<th>2 kHz</th>
<th>4 kHz</th>
<th>8 kHz</th>
</tr>
</thead>
<tbody>
<tr>
<td>Q</td>
<td>8</td>
<td>11</td>
<td>18</td>
<td>22</td>
<td>18</td>
<td>45</td>
<td></td>
</tr>
</tbody>
</table>
column speakers

- reducing the extreme frequency dependency of Q by
  - segmentation and tilting
  - each chassis gets individual signal
electronically steered column speakers: Line arrays

- column speakers with
  - individually steered chassis (delay, frequency response)
- focused radiation in the vertical plane
- horizontal directivity determined by chassis characteristics
- advantages compared to conventional column speakers:
  - beam orientation adjustable by electrical means → vertical mounting possible
  - radiation characteristics can be optimized (frequency response)
electronically steered column speakers: Line arrays

- sound field of a row of point sources:
  - 16 chassis
  - 0.5 m separation
  - 500 Hz
  - all chassis in phase
electronically steered column speakers:

**Line arrays**

- sound field of a row of point sources:
  - 16 chassis
  - 0.5 m separation
  - 500 Hz
  - each chassis with individual delay
electronically steered column speakers: Line arrays

- behavior of a row of point sources:
  - focusing by interference
  - line source distance dependency with approx. -3dB/distance doubling
  - limitation to low frequencies: focusing vanishes for array length $< \text{half first Fresnel zone}$
  - limitation to high frequencies: spatial aliasing for chassis separation $> \lambda$
line arrays: practical aspects

- high frequency horns to avoid necessity of many tweeters close to each other
- delay → adjustment of beam orientation (several simultaneous beams are possible!)
- apply frequency filtering (outer chassis radiate low frequencies only) → reduction of strong frequency dependency of Q
- bending of column at the lower end for an improved near-field supply (J-form)
line arrays: example
suppression of feed-back
wobbling
wobbling

- frequency modulation → permanently altering phase relation in the feed-back path
- original idea: 1928 Zwicker:
  - loudspeaker or microphone suspended on a oscillating pendulum
- digital solutions:
  - optimal modulation frequency: 4.5 Hz
  - optimal frequency variation: ± 5 Hz
  - gain: 5 dB
- problem: audible at low frequencies
frequency shifting
frequency shifting

- microphone signal is shifted in frequency by $\Delta f$ prior to radiation

- consequence
  - smoothing of the peaks in the room transfer function

- digital solutions:
  - typical shift $\Delta f$: 5 Hz
  - gain: 6 dB
notch filter
notch filter

- for fix microphone and loudspeaker positions relatively few frequencies show possible instability
- suppression of these frequencies by narrow banded notch filters
- typical bandwidths 5 Hz
- for moving microphone positions: adaptive filters
estimation and compensation of the feed-back path
estimation and compensation of the feed-back path

example: subtraction filter:

- estimation of the feed-back path
- electronic simulation of the feed-back path by a digital filter
- subtraction of the signal that passed through the feed-back loop once
estimation and compensation of the feed-back path

\[ \text{G}_1: \text{feed-back path} \]
\[ \text{G}_1': \text{estimate of G}_1 \]
estimation and compensation of the feed-back path

- difficulty: estimation of feed-back path has to be accurate with respect to phase
  - systems are often time-variant (variable geometry, changing air temperature ...)
  - promising strategy: continuous measurement, e.g. with MLS at inaudible level
speech intelligibility
speech intelligibility: introduction
speech intelligibility: influencing factors

- signal/noise ratio
- reverberation time
- ratio of direct and diffuse sound (determined by total absorption and distance source - receiver)
- early reflections (comb filter effects for time of arrival differences of about 1 ms)
- late reflections (echoes for time of arrival differences of about 50 ms)
speech intelligibility: evaluation

- **subjective measure**
  - percentage of syllables correctly understood → intelligibility of syllables

- **objective measure**
  - Articulation Loss: ALcons
  - Speech Transmission Index: STI
  - Rapid Speech Transmission Index: RASTI
  - Speech Transmission Index PA: STI-PA
  - Deutlichkeitsgrad: D50
intelligibility of syllables
intelligibility of syllables

- basis of all measures
- subjective evaluation with speaker and panel of test persons
- determination of the ratio of correctly understood syllables
- conversion tables to translate intelligibility of syllables into intelligibility of words and into intelligibility of sentences
articulation Loss
articulation Loss

- speech intelligibility measure for prognosis:

\[
%AL_{\text{cons}} = \frac{200D^2RT^2N}{V \cdot Q}
\]

for \( D < 3.2r_H \)

\[
%AL_{\text{cons}} = 9RT
\]

for \( D \geq 3.2r_H \)

\( D \): distance source-receiver

\( RT \): reverberation time in range 500 Hz...2 kHz

\( V \): room volume

\( Q \): Q factor of loudspeaker

\( N \): ratio of loudspeaker power that contributes to diffuse field and to direct sound

\( r_H \): critical distance = \( \sqrt{\frac{QA}{16\pi}} \)

\( A \): total absorption
articulation Loss

- with help of $\%AL_{cons}$ formula $\rightarrow$ dimensioning of a PA system

- significance of $\%AL_{cons}$ values:
  - $< 10\%$ $\rightarrow$ very good speech intelligibility
  - $10\ldots15\%$ $\rightarrow$ good speech intelligibility
speech transmission index
speech transmission index

- speech intelligibility measure for measurements and calculations
- simulation of speech signals by slowly amplitude modulated noise (modulation depth 100 %)
- calculation of intelligibility from resulting modulation depth at receiver
speech transmission index

- influence on modulation depth:
  - disturbing noise
  - reverberation
  - echoes
  - interference of several sources
speech transmission index

- measurement is possible without synchronization between sender and receiver → important advantage!
- variations:
  - RASTI
  - STI-PA
- very good intelligibility for:
  - STI, STI-PA, RASTI > 0.85
Deutlichkeitsgrad D50
Deutlichkeitsgrad D50

- speech intelligibility measure for measurements and calculations

- concept:
  - evaluation of precedence effect: signal energy within the first 50 ms after direct sound is helpful, later components are detrimental
  - calculation of the early/late energy ratio from the impulse response $h(t)$

\[
D50 = \frac{\int_0^{50ms} h^2(t)dt}{\int_0^{\infty} h^2(t)dt} \times 100\%
\]
Deutlichkeitsgrad D50

- calculation:
  - room acoustics simulation programs
- measurement:
  - impulse response measurement, e.g. with MLS
- very good intelligibility for D50 > 50%
localization
> due to the precedence effect there is chance to localize the original source and not the loudspeaker

- echo
- inaudible
- and correct
direct sound localization
localization

- dimensioning:
  - direct sound of original source has to arrive first at listener
  - installation of loudspeaker behind microphone but increased feed-back tendency
  - introduction of electronic delay
types of PA systems
centralized PA systems
centralized PA systems

- all loudspeakers located at one single position
  - horn speakers
  - speaker clusters
centralized PA systems

- advantages:
  - relative homogeneous level distribution can be achieved
  - localization controllable

- disadvantage:
  - is not working in halls that are too reverberant (RT > 2 sec)
distributed PA systems
distributed PA systems

- many distributed loudspeakers
  - mounted e.g. in ceiling
  - distributed column speakers (often seen in churches)
distributed PA systems

- **advantage:**
  - works in reverberant rooms and in rooms with low ceiling

- **disadvantages:**
  - inhomogeneous level distribution
  - possible problems in areas with supply from two different speakers (interferences)
  - localization seldom o.k.
summary: principles of a well designed PA system for speech

- high direct sound level
  - loudspeakers with high Q
  - small distances loudspeakers - listeners
summary: principles of a well designed PA system for speech

- low disturbing noise level
  - high transmission loss to outside sources
  - noise control of sources in the room itself (air conditioning system, seats, ...)
summary: principles of a well designed PA system for speech

- low diffuse sound level
  - short reverberation time
  - number of loudspeakers as small as possible
  - loudspeakers with high Q
  - orientation of loudspeaker main radiation direction towards audience areas
  - attenuation of bass frequencies
    - not relevant for speech intelligibility
    - higher reverberation times at low frequencies
    - Q of loudspeakers is usually small at low frequencies
summary: principles of a well designed PA system for speech

- avoid echoes
  - room acoustics:
    - no concave reflecting surfaces
    - avoid unstructured and reflecting parallel walls
    - suitable amount of absorption
summary: principles of a well designed PA system for speech

- avoid interferences
  - low number of loudspeakers
  - careful design of zones that get sound from two different loudspeakers
summary: principles of a well designed PA system for speech

- **homogeneous level distribution**
  - large distances loudspeaker - listeners
  - "tuned" loudspeaker arrays (line-arrays)
summary: principles of a well designed PA system for speech

- high maximal amplification
  - small distance speaker - microphone
  - directional microphones
  - large distance loudspeaker - microphone
  - focusing loudspeakers
  - no unnecessary open microphones → intelligent mixers
PA systems for music
PA systems for music: elements

- microphones and instrument pickups
- mixing consoles
  - main mix
  - monitor mix
- signal processing units
  - equalizers
  - compressors
  - various effects
- power amplifiers and loudspeakers
  - main mix $\rightarrow$ audience
  - monitor mix $\rightarrow$ performers on stage (or alternatively in-ear monitoring)
# PA systems for music: necessary power

<table>
<thead>
<tr>
<th>performance type</th>
<th>W/person</th>
</tr>
</thead>
<tbody>
<tr>
<td>pop/light rock band <em>indoor</em></td>
<td>8</td>
</tr>
<tr>
<td>hard rock or metal band <em>indoor</em></td>
<td>12</td>
</tr>
<tr>
<td>pop/light rock band <em>outdoors</em></td>
<td>16</td>
</tr>
<tr>
<td>hard rock or metal band <em>outdoors</em></td>
<td>24</td>
</tr>
</tbody>
</table>

*Note: W/person values are approximate and may vary depending on the specific requirements of the event.*
PA systems for music: loudspeaker configurations

<table>
<thead>
<tr>
<th>power</th>
<th>configuration</th>
</tr>
</thead>
<tbody>
<tr>
<td>10 kW</td>
<td>2 subwoofers and 2 mid/high speakers</td>
</tr>
<tr>
<td>20 kW</td>
<td>4 subwoofers and 4 mid/high speakers</td>
</tr>
<tr>
<td>50 kW</td>
<td>special arrangements: subwoofers + line arrays</td>
</tr>
</tbody>
</table>
PA systems for music: open-air: annoyance in the neighborhood

- open-air concerts need assessment of possible annoyance in the neighborhood
- VDI standard → estimation of acoustically emitted power for various types of performances
eth-acoustics-2