Tu. 2/5: Ch 5 *Intensity and Measurement* **Th. 2/7:** Ch 15 *Room Acoustics*	HW5: Ch5: 1^w,3^w,6^w,11^w,12,14^w Ch15: 2,3^W,4^W,7,8^W,12^W	**Mon. 2/4** or **Tues. 2/5:** Lab 5 *Diffraction & Attenuation*
Tu. 2/12: Review **Th. 2/14:** Exam 1 (*Ch 1,2,3,4, 5,15*)	**Tues – come with questions** presentation proposal	**Mon. 2/11** or **Tues. 2/12:** Lab 6 *Room Acoustics*

Equipment

- **Wave tank applet - http://www.falstad.com/ripple/**
- **Homework for next Tuesday –** look over online review material (and homework, labs, class notes) and come in with questions
- **Laptop with Odeon & dome simulation**
- **vPython dome**
- **Dome slides from Beautiful data**

Administration

- *Pr. 7, you'll want to look-up "phoneme" in the book – essentially, distinct sounds of speech.
- What do they want for the review? I was thinking of going over homework problems that they requested. – I've posted review questions and chapter notes to the course website (address at the top of the syllabus)

Chapter 15: Room Acoustics

Introduction: Over the last few days we've focused on how sound propagates, how its intensity decays as it travels out, how it reflects off surfaces (and adsorbs into them), and how it bends around obstacles. We've even applied this understanding to describe simple structural acoustic - the curve of a band shell. Now we are going to get more specific in structural acoustics. The goal is to understand how the shape and material of a room affect the listening experience. We'll first categorize the qualitative properties of a listening experience on the one hand and the quantitative components of a sound. Then we'll talk about how the environment can be manipulated to sculpt the sound components to yield a quality listening experience.

15.1 General Criteria For Room Acoustics

Motivation: Bad listening Experiences

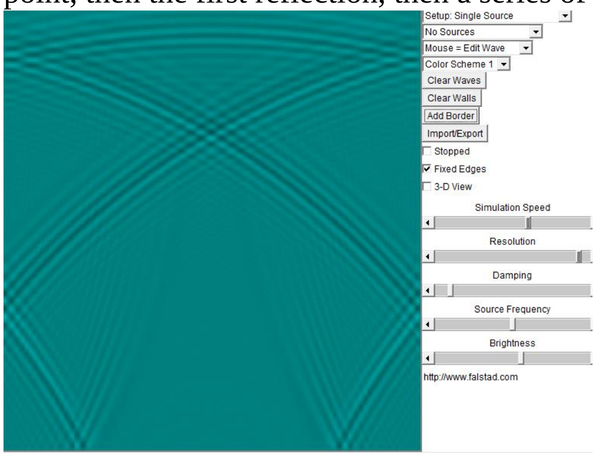
- **Q:** Think about particularly *bad* listening experiences you've had. Perhaps in your car, in a class room, in a chapel, in a theatre, in a concert hall, in a club,… None too technically, what was bad about it?

- **Listening Qualities**
  - **Clarity** (easily distinguish distinct notes, crisp sounds remain crisp)
  - **Uniformity** (no dead spots, one spot isn't blasted away while the other is dead silent)
  - **Envelopment** (audience members should feel like they are truly present in a place of music, not just sitting in front of a music source, the sound washes over them, they are in its midst's, over whelmed, soothed, rocked, by it.)
  - **Reverberation** (it's nice to have the sounds smoothed and enhanced a little, let the sound last a time, at an appropriate level, decay at a reasonable rate. – balanced against the desire for clarity and for (no Echoes)

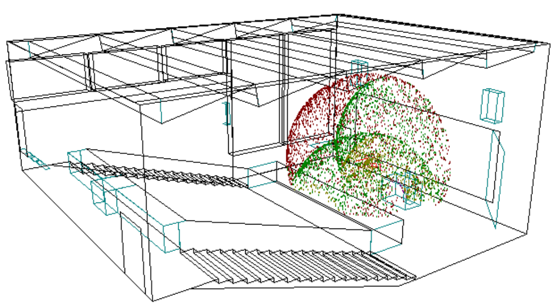
  - **Free from Echoes** (though sound has to bounce off of walls to come at you from other directions, it can't take so long as to sound like a fainter, distinct event, confusing, cluttering)
  - **Performer satisfaction** (doesn't just have to be good for the audience, must be good enough for the performers so that they can hear each other appropriately, not be distracted by echoes, so that the sound is generally pleasing to them, so they can respond to it appropriately.)
  - **Free from noise** (if in the middle of a p (60dB) passage, city traffic, at 70dB, had better be blocked out.)

- **Sound Components**
  - So there are the demands, what is the system, what are we working with…

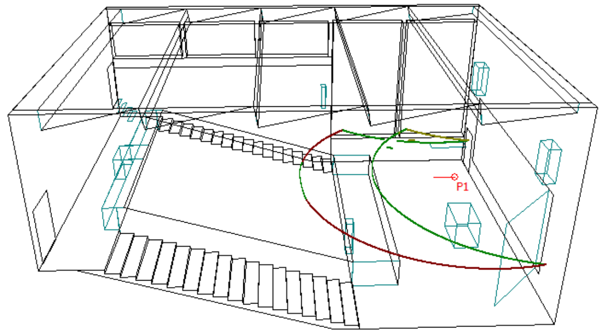
Demo: Ripple tank app with walls, no source. Send in just one pulse, see it wash over a point, then the first reflection, then a series of ever weaker reflections.

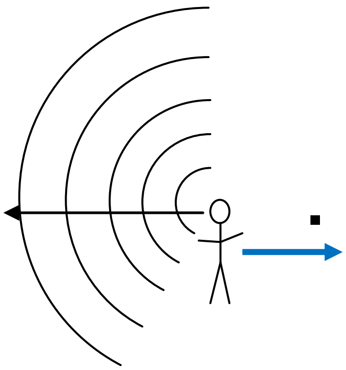


Demo: Odeon Auditorium, 3-D pulse



  - **The individual pieces**
    - **Direct sound** – (draw & show in Odeon – no diffusion, simple xy plane source) straight from the instrument to your ear
      - **Arrival Time:** If the listener sits a distance r away from the sound source, then we can use $v_S = r/t$ to find the how long after the first wave front is emitted, until it arrives at the listener: $t = r/v_s$ where v_s is roughly 344 m/s. For someone sitting mid-audience in this simulation, it takes about _22__ms for the first bit of sound to wash over them.

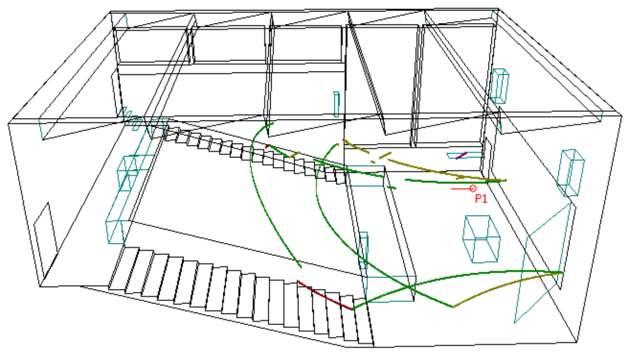


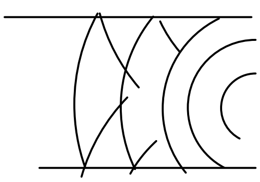


-
- **Intensity** We saw last time that the intensity drops off with distance. $I = \dfrac{P}{4\pi r^2}$

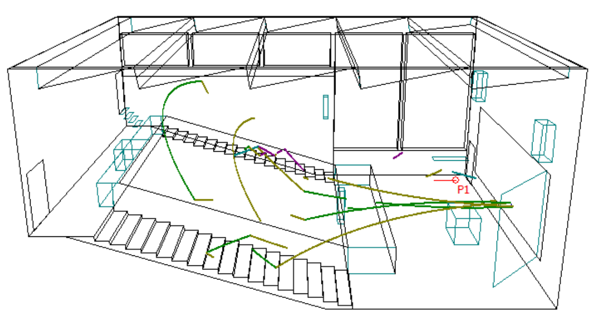
▪ **Early reflections** – (draw & show in Odeon) off of a wall / ceiling to you.

- **Arrival Time:** The path this sound front must travel could be two or three times the direct path; thus it would take twice to three times as long to reach you. In the simulation, it's not until __27__ms that the reflections from the walls wash over the person.



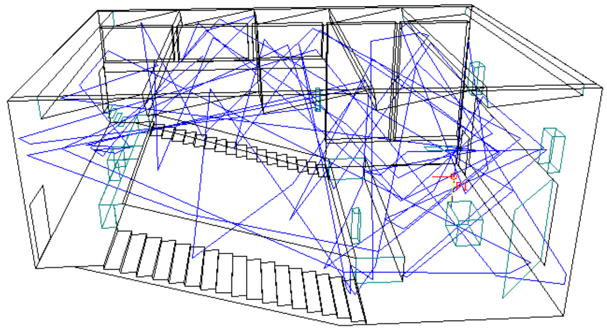


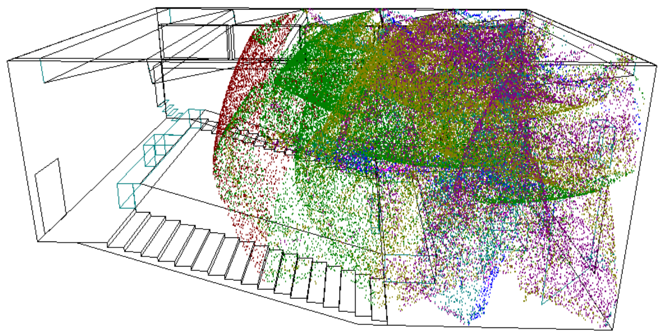
-
- **Intensity:**
  - ○ **Decay with distance:** The intensity of the sound decreases like the square of the increase in distance, Twice the path length, $1/4^{th}$ the intensity.
  - ○ **Decay with reflection**: When a sound wave front splashes up against a surface, some of its oomph is passed into the surface and the balance is reflected back again. A reflected wave is less intense than the incident one. Odeon color codes the sound front to show how many times it's reflected, which gives a sense of how much it's weakened.



1. 36 ms later

- **Reverberant sound** – (draw & let Odeon run free) Think of a complicated pool shot '…off the far bumper, then the side bumper, off the 9 ball, back off the side bumper, and then into the 8 ball…'



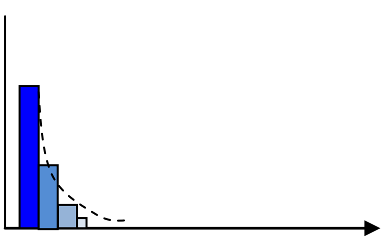


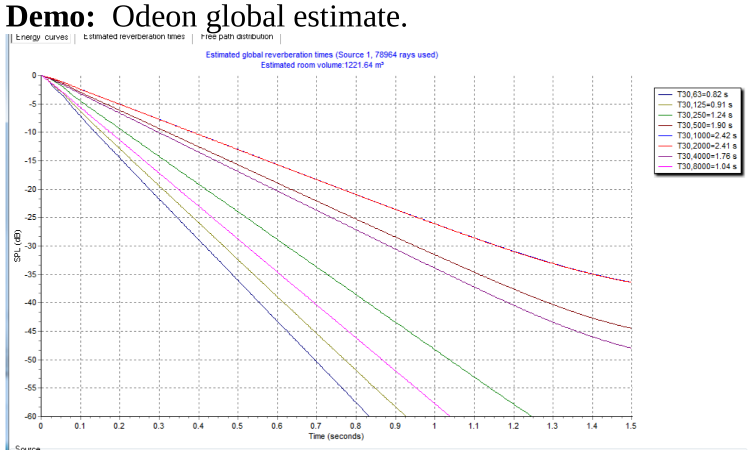
A sound front may bounce off of many surfaces in its course from the source to your ear. There are an infinite number of paths that could be taken, of an infinite number of reflections and path lengths.

- **Arrival Time:** There is a continuous stream of reflected waves making it too your ear. Each one having taken a longer, more circuitous path; thus, each one taking a longer time to arrive to you.
- **Intensity:** The longer the path, the weaker the intensity; the more reflections, the weaker the intensity.

Tracking an individual morsal of a sound pulse, it gets weaker and weaker as time goes on.

Demo: Odeon global estimate.





- **Putting these together**

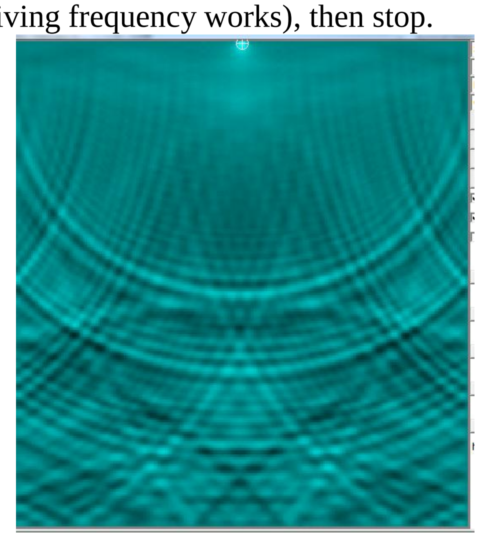
Demo: Ripple tank app with wall**.** Again, send in just one pulse, see it wash over a point, then the first reflection, then a series of ever weaker reflections.

- This is just for one wave front & its family of reflections. Getting closer to music, let's think of a sustained note: a series of wave fronts, each with their family of reflections.

- **Reverberation time 1: Specific Example**
  - Let's walk through a simple, specific example and see how these pieces fit together to give you a single, composite sound.
  - Let's say that the direct sound has to travel 3.44 meters, the 1st reflected sound travels twice that, the 2nd reflected sound travels 3 times that,… To make it simple, we'll just consider the diminishing effect of reflection, we'll say that the surfaces are wood and reflect only 30% of the sound intensity (we'll get into different materials later)
    - **Direct wave**
      - Arrival time: $x/v_s = t_d = 0.01s = 10$ ms.
    - **1st reflection**
      - Arrival time: $2x/v_s = t_{r1} = 0.02s = 20$ ms.
      - Intensity: $0.3\ I_{Direct}$
    - **2nd reflection**
      - Arrival time: $3x/v_s = t_{r1} = 0.03s = 30$ ms.
      - Intensity: $0.3*0.3\ I_{Direct} = 0.09\ I_{Direct}$
    - **3rd reflection**
      - Arrival time: $4x/v_s = t_{r1} = 0.04s = 40$ ms.
      - Intensity: $0.3*0.3*0.3\ I_{Direct} = 0.027\ I_{Direct}$

Sustained Source – Say someone plays and holds a note, then you can imagine wave after wave washing over you.

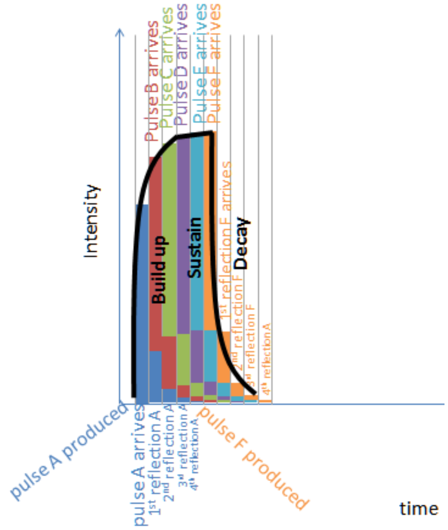
Demo: Wave tank with walls. Send in a series of pulses (oscillating the driving frequency works), then stop.



- **Build-up:** The intensity of the sound you hear builds up as first the stream of direct waves, then the stream of 1st reflection waves, then the stream of 2nd reflection,… arrive. However, each contribution is weaker than the last. The intensity exponentially approaches a maximum.
- **Sustain:** If the note is sustained long enough, the net intensity maximizes.
- **Decay:** When the performer stops, first the direct sound stops, then the first reflected, then the 2nd,…So the sound decays exponentially.

(ripple tank doesn't have strong enough damping / adsorbant enough walls, closest I can get to look right is, when stopped, clear source and walls, then resume.)

- As long as the note is being sustained, these contributions layer on top of each other to give a total sound intensity.



(sketch the **build-up** and **sustained** parts of Plot 15.3 b)

- Once the source stops, the bottom falls out of the house of cards, and each consecutive reflection dies off, in order
- First the first reflection stops coming to you, then the second…

(sketch the **decay** part of Plot 15.3 b)

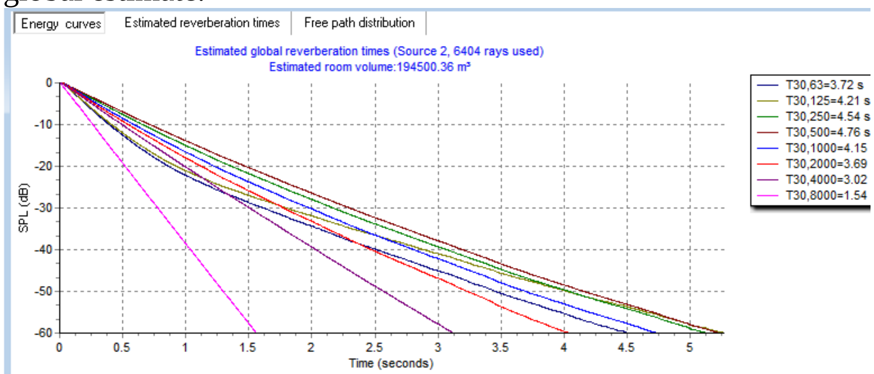
Demo: SLM & Logger Pro – clap and see the plot of SIL.

- **Reverberation Time, T_r = T_{-60dB}**
  - We characterize the growth and decay of a sound by the time it takes for the sound intensity to decay by a factor of one million, 1,000,000, *six* powers of 10, i.e., by 60 dB, This is a convenient level to use since that's when even the loudest sounds of an orchestra: *ffff* decays away to the level of the quietest: *pppp* –all intents and purposes, even the loudest sound has died.
  - **Motivation: Reverb. time is important.** In a moment, we'll see that this quantitative measure of a sound's duration relates directly to many of the qualities of a listening. So we'll take a moment and think about what properties of the environment determined it.
  - **Q:** think about sound in the chapel, then think about sound in this room. Which would you say lasts longer, which has the longer reverberation time?
    - **A:** The chapel.
  - **Why?**
    - Partly because it's **bigger**. The reverberation time depends on how long it takes for each step in the intensity – one reflection, then the next, then the next… That depends on the time of flight. Well, the speed of sound is the same in here and in the chapel, but in the chapel, sound generally has a lot further to go before it hits a

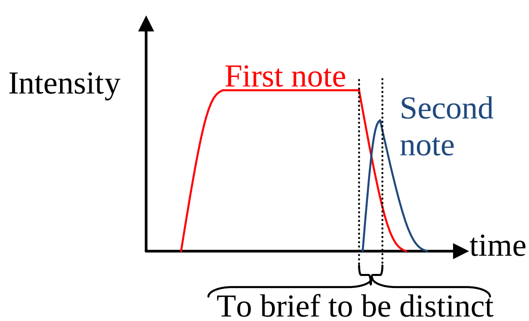
wall, and then hits you – the bigger a room, the bigger time lag between the reflections.

- Partly because it's **more reflective.** The walls, ceiling, and floor of that room reflect a larger percent of the sound.

- **Demo: Odeon -** Copenhagen Central Station source 2 – Hall, global estimate.



- **How do these effect the listening qualities?**
  - ***Clarity*** – If you're playing a quick passage of music in a room with a long reverberation time, you can imagine the sound you *just* made still reverberating and being heard while you're on to saying/playing the next sound. For the listener, the sounds would blend into each other – not be very distinct. With some music, maybe you want to provide a wash of sound, but when you're speaking, you probably want people to hear the distinct sounds and understand what you're saying. Thinking of the plot we'd build of sound slowly growing, sustaining, and then decaying, here's what such a plot might look like if you sustained one long note and then played one quick one – so quick that you're done playing it before the first sound has really decayed away.



  - How brief is too brief a sound / how long is too long a reverb time? A good rule of thumb is that you want your old sound to decay to about 1/10th its full intensity before you finish with your new sound – so at least for a little while, the listener pretty much just hears the new sound on its own. If Tr is the time to decay by 60dB, i.e., by 6 orders of magnitude (to 1/1000000th its full strength), then 1/6 T_r is about the time it takes to decay to 1/10th its full strength. So, you want
    - $\Delta t_{sound} > \frac{1}{6} T_r$

Demo: "Good-morn-ing" (like Pr. 7)

One student says Goooood, one says moooorn, and one says iiiiinnggg.

We will vary ~ decay times. I point with my right hand – you start, I point with my left, you stop. Play around with relative starting and stopping times. If the time interval between phonemes or syllables starting is small compared to the 1/6 T_r (time to decay by 10dB, i.e. time for sound to decay to 1/10th its original intensity), then things get muddled. You'll play with this in Pr. 7

Phoneme = distinct sound. For example, "sound" has 5 letters, 4 phoneme's, and 1 syllable. For real clarity, you need to be able to distinguish each *sound.*

    - most clear, least confusion in the noise, strongest direct signal relative to Early reflections and reverberant sound. Early reflections and reverberant sound should come and go very quickly, relative to the rate of articulation – how frequently a new, distinct sound is produced. So, the room is built to bring people up front, ensure direct line of sight. Reverberation can be sacrificed by using adsorptive material to cut out the sound after fewer reflections, in a shorter time.

  - ***Uniformity*** – again, bring everyone up near the front. Also avoid hot spots of reflected sound, no capital building like dome to focus reflected sound waves.
    - **Demo:** Vpython of our dome.
  - Then again, no regular, parallel walls to allow sound to bounce back and forth and back and forth, like in an organ pipe. Instead, irregularly shaped walls, or big tiles at odd angles to bounce sound in all directions.
  - ***Envelopment* --** want the reflected sound to come from all around. The earlier this is in the path of reflections, the more intense it is, and so the more the sound seems to envelop. Again, irregular surfaces are good, but note, the reflected sound should be much less than the original, otherwise the reflected sound will be perceived as coming from another source / an echo.
  - ***Performer satisfaction*** – must have good reflection to them too, not dead, and not too loud a reflection, no parallel hard walls, front and back, or left and right (these can support a flutter echo bouncing back and forth)

- **Different desires for different types of sound: The Church dilemma – Organ vs. Oration**:
  - **Organ** music has a very large, enveloping sound, typically want the *warmth* of notes gliding into each other – want *large reverb time* 1.5 – 2.5 sec., could be up to 8 sec. In a large, stone cathedral! Hard brick walls, no drapes.
  - But the **pastor** wants to be *understood, clarity*. He/she too may appreciate some reverb. These are weighty subjects and the words should have weight and size and power. But they must also be perceived as words. *Short reverb time.*

- **How is this handled?**
  - **Soften surfaces**
    - Maybe there are large banners hanging on the walls, acoustically soft and sluggish, adsorb sound. For an organ performance, they may be taken down.

- **Example: Carleton Concert Hall**

- **Movable ceiling panels** (fig 15.5) – it addresses this kind of problem with movable ceiling panels and wall panels. Ideally, they would lower the panels for a debate and raise them for a concert.
- Sound bounces off of them just as do balls off of a floor, reflect in the direction perpendicular to the floor, and just keep going in the direction parallel. So equal angles of incidence and reflection. What do these curved panels do? Shorten time, also redirect sound to the back..

15.3Reverberation Calculation

- **Reverberant time** = time in which sound level drops 60dB below original level.
- We can measure reverberation time by seeing how rapidly a clap's sound decays or we can simulate it as in Odeon. But there's a simpler way to get an estimate.
- $T_r \approx 0.16\,{}^{s}\!/\!{}_{m} \cdot \frac{Volume}{Area_{effective}}$: Bigger room, longer time between reflections, longer reverb time, more surface area means more reflection, more diminishing of sound – damps out more, lower reverb time.
  - $Area_{effective} = \alpha_1 A_1 + \alpha_2 A_2 + ...$
    - A1 = surface area of wall, seat, etc.
    - α_1 = adsorption coefficient, table 15.1, for that surface, the fraction of the sound intensity that gets adsorbed. Note: an open door or window adsorbs *all* the sound, so $\alpha = 1$. (important for HW problem)
    - An effective area says, having a soft blind that adsorbs 1/10th of the sound is the same as having a wide open window, that adsorbs all the sound, but is 1/10th as big.

- This is a fairly easily quantified parameter, and one easily manipulated, so we focus on how it can be adjusted to optimize a building's acoustics.
- Note: different materials respond differently to different frequencies.
  - **Demo:** Odeon Materials list – see different adsorptivites.
  - Which means a space has different reverberation times for different frequencies.
    - **Demo:** Odeao – estimate reverb time.

Example like 12: Calculate reverb time for a room. Judge its quality. Do an example T_r calculation for 500 Hz tone in a typical dorm room

Quantities

$A_{floor} = 15.7\ m^2$	$\alpha_{floor} = 0.03$ (vinyl)
$A_{Ceiling} = 15.7\ m^2$	$\alpha_{ceiling} = 0.06$ (painted concrete)
$A_{window} = 2.2\ m^2$	$\alpha_{window} = 0.2$ (glass)
$A_{door} = 4\ m^2$	$\alpha_{door} = 1.0$ (open)
$A_{Walls} = 4 * 14.5\ m^2 - A_{Window} - A_{door} = 51.8\ m^2$	$\alpha_{wall} = 0.06$ (painted concrete)
$Volume = 57.4\ m^3$	

Relations

$$T_r = 0.16\,{}^{s}\!/\!{}_{m} \cdot \frac{Volume}{Area_{effective}}$$

$$Area_{effective} = \alpha_1 A_1 + \alpha_2 A_2 + ...$$

Algebra / Numbers

$$T_r = 0.16\,{}^{s}\!/\!_{m} \cdot \frac{Volume}{\alpha_{window}A_{window} + \alpha_{walls}A_{walls} + \alpha_{floor}A_{floor} + \alpha_{ceiling}A_{celiling} + \alpha_{door}A_{door}}$$

$$T_r = 0.16\,{}^{s}\!/\!_{m} \cdot \frac{57.4m^3}{0.2\cdot 2.2m^2 + 0.06\cdot 51.8m^2 + 0.03\cdot 15.7m^2 + 0.06\cdot 15.7m^2 + 1.0\cdot 4m^2} = 1.49s$$

This may seem long, but it's for a *bare* room, like a bathroom. Toss in a rug, curtains, a bed with blankets, things with adsorptivities of 30 – 60%, and the reverberation time comes down dramatically.

15.4Reverberation Sound Levels

- Reverberant Sound Level and Radius
- **Reverberation stronger than direct.**
  - You expect that since the Direct sound dies off like 1/r2, while the reverberant keeps criss crossing, new, strong added to old decayed, there would be some point at which the single contribution of the Direct sound would die to weaker than the many weak contributions to reverb sound. This happens out at a radius of 0.06 sqrt(V/Tr).
- **Direct Weaker but Earlier.**
  - An interesting trick happens in the brain though. Rather than perceiving the sound to come from the sides and behind and above, as is the dominant, reverb sound, we still *perceive* the source to be in front of us! Because it *got to us first*. When reverb is far stronger than the initial, direct sound, then this changes.

15.5Sound Reinforcement

- If you can't bounce the sound off of surfaces to get it where you want it to go, you just wire it in. Have a microphone that responds to the sound, wired to an amplifier to increase the electrical response, then to a speaker somewhere else.
- catastrophic positive feedback with bad placement
- Every wave through its own medium has its own wave speed. Electrons in a wire produce a wave much faster than molecules in the air – much more rigid restoring force and much less massive. So if we do this, over a long distance, the sound will get to us much much sooner than it should. The wired sound gets to us before the direct sound does, and we may perceive the sound source to be the speaker, not the real source. A similar thing: you're on the phone to someone across town, lightning hits over at their side, you hear it in the background over the phone, then you hear it through the air.

15.6Spatial Perception (hold off until we talk about the ear)

- Cues, reverb time gives us size of room
- Where the sound comes from first gives us location
  - How de we determine where it comes from at all?
    - High frequency diffraction around our heads, reflection off our shoulders. Good for front and back, top and bottom, we are very asymmetric in this way, but left right, we are quite symmetric.
    - Moderate to High frequency, which ear gets the stronger signal.

- Low frequency, which ear gets it first. – For a constant tone, this is a phase shift.