Sound Systems Design and Engineering

Based on lectures by Dan Perelstein

Notes taken by Daniel Moore

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These notes are not endorsed by the lecturers, and I have modified them (often significantly) after lectures. They are nowhere near accurate representations of what was actually lectured, and in particular, all errors are almost surely mine.

Contents

Part I

Loudspeakers

1 System Design Building Blocks

1.1 Decibel Math

The decibel (dB) involves the use of a logarithm to allow large differences and numbers to be written as conveniently small numbers. A decibel is a ratio between two pressure levels (ie. the measure of deviation of air at a certain point due to sound). These get big so quickly that logarithms quickly become necessary.

There are two definitions for decibels:

$$\text{dB} = 10 \log_{10}\left(\frac{A}{B}\right)$$
$$\text{dB} = 20 \log_{10}\left(\frac{A}{B}\right)$$

The first definition is in reference to power, the second in reference to voltage. In this class, we'll only work with the second one, as that's the one that is generally useful for sound. The first one is useful when talking about power amplifiers. For the future, we'll write log instead of $\log_{10}$ for the sake of simplicity.

A is the pressure value we want to measure, and B is the reference value. The reference value we use (B) is the quietest sound that it's possible to hear.

1.1.1 Logarithms (and their Applications)

A base-10 logarithm like we'll be using answers the question "what exponent does 10 need to be raised to to equal our value?" For example, $10^1 = 10$, so

$$\log(10) = 1$$

This asks the question "what exponent does 10 need to be raised to to equal 10," for which the answer is 1. Another example, $10^0 = 1$, so

$$\log(1) = 0$$

This asks the question "what expoent does 10 need to be raised to to equal 1," for which the answer is zero.

Logarithms convert multiplication into simple addition and division into simple subtraction. Written out, what this means is that we can write $\log(AB)$ instead as

$$\log(AB) = \log(A) + \log(B)$$

Similarly, we can also write $\log(A/B)$ as

$$\log\left(\frac{A}{B}\right) = \log(A) - \log(B)$$

There's another important thing to note about logarithms. If we measure the sound pressure level (SPL) at some point to be x dB, that means that we know

$$x = 20\log\left(\frac{A}{B}\right)$$

How do the number of decibels change for a sound twice as strong?

$$\begin{aligned}
20\log\left(2\frac{A}{B}\right) &= 20\log(2) + 20\log\left(\frac{A}{B}\right) \\
&= 20\log(2) + x \\
&= 20(.03) + x \\
&= 6\text{dB} + x\text{dB}
\end{aligned}$$

What this means is that any time we double the pressure of the sound, we're adding 6 dB.
Let's see what happens if we do this but we half it instead:

$$\begin{aligned}
20\log\left(\frac{1}{2}\frac{A}{B}\right) &= 20\log\left(\frac{A}{2B}\right) \\
&= 20\log\left(\frac{A}{B}\right) - 20\log(2) \\
&= x\text{dB} - 6\text{dB}
\end{aligned}$$

So, halving the pressure level means we subtract 6 dB.
One more example: what happens if we have an SPL ten times as large?

$$\begin{aligned}
20\log\left(10\frac{A}{B}\right) &= 20\log(10) + 20\log\left(\frac{A}{B}\right) \\
&= 20(1) + x \\
&= 20\text{dB} + x\text{dB}
\end{aligned}$$

This means that to get a sound with ten times the pressure, we have to add exactly 20 dB.

1.2 Inverse Square Law

As an introduction, the surface area of a sphere is

$$\text{Surface Area} = 4\pi r^2$$

where r is the radius of a sphere. Note that if we double the radius, there will be 4 times as much surface area.

The inverse square law isn't just a sound thing—it's common between sound, gravitational pull, electric field strength, radiation, etc. If we have a point source that obeys the inverse square law (imagine, for example, a speaker in the air broadcasting sound equally in all directions), the intensity of a sound is inversly proportional to the distance from the source (as distance increases, intensity decreases, and visa-versa). Specifically, they're inversely proportional with a square relationship (hence the name). For example, if we move away from the source a distance of 2m, the intensity drops to 1/4 of what it was. This is because the sound is being projected in a sphere, and as we get farther away, the surface area is increasing, but the total sound pressure isn't increasing, so the sound has to spread out more to cover the area.

As an example. If a speaker 1 ft away from you has volume x. At 2 ft, it would have volume of $\frac{1}{4}x$. At 3 ft, $\frac{1}{9}x$. At 4 ft, $\frac{1}{16}x$. And so on. We get these numbers by squaring the distance and putting it on the bottom.

An important conclusion from this: if you're using surround sound (ie. sound from behind the audience), you'll get more even coverage if you put the speakers farther behind the audience. If the speakers are, for example, 1 ft behind the back of the audience, the volume even 6 feet away (one or 2 rows up) will be already $\frac{1}{36}$ the intensity since it's 6 times as far. On the other hand, if we put the speakers 12 ft away from the last audience member, the volume 6 feet away (one or 2 rows up) is only 1.5 times as far away, so it's only $\frac{4}{9}$ as quiet (if I'm doing my math right). This still isn't the best, but it's a lot better. We can also use vertical height to increase the distance from the audience without having to go as far back.

1.3 Reference Signals

1.3.1 Sine Waves

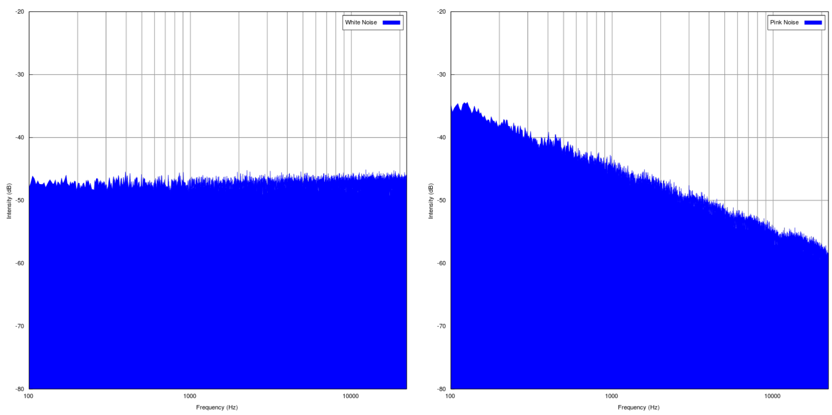
Sine waves are patterns of air pressure where the pressure in some direction can be modelled using a sine curve (very generally, this means that areas of high pressure happen at regular, predictable intervals at regular, predictable maximum intensities, and areas of low pressure similarly happen at regular, predictable intervals and with regular, predictable minimum intensities).

1.3.2 Fourier

Fourier found in the 1820s that any signal, no matter how complicated, can be written as the sume of sine waves at different amplitudes and at different phase relationships. Through Fourier analysis, we can convert from looking at a signal as a function of time, to looking at a signal as a function of frequency. Theoretically, we could totally replicate anything we hear or record with a sum of sine waves, but that would take literally tens of thousands of sine waves every moment, so it's hard.

1.3.3 Noise

There's two types of reference noise we use: white noise and pink noise. White noise has more audible high end than pink noise, and pink noise has more audible low-end. White noise has equal amplitude at every frquency. Pink noise takes white noise and applies a "pinking" filter, which takes away 3 dB per octave. There are more frequencies at a given octave at high frequencies, so in white noise, the high frequencies have much, much more energy, making them overpower the lower octaves. The goal with pink noise is to give us something that sounds like it's evenly distributed betwen notes. Here's the difference between white and pink noise (respectively) on a frequency spectrum:



2 Loudspeakers

2.1 An Introduction to Loudspeakers

Speakers are transducers: electronic components that convert one type of signal to another. Specifically, speakers convert signals from electrical energy to mechanical/acoustic energy.

Ideally, we want to be able to produce every sound that humans can hear (20 Hz to 20 kHz) so we can recreate the sounds we're trying to represent as accurately as possible. This is usually not really possible

2.2 The -6 dB Point

Remember that -6 dB represents halving the sound pressure level (SPL) of a signal. The width that the manufacturers give for a loudspeaker is generally going to e the point at which the volume is 6 dB lower than it is at the same distance on-axis. These levels are measured at 1 kHz, since the high-frequencies attenuate at different rates than low frequencies when you're going off-axis. If we want even coverage, we want to line up the 6 dB point of one loudspeaker with the 6 dB point of the other. If we overlap too much, we'll get areas with more than 0 dB, and if we separate them too much, we'll get areas with less than 6 dB.

3 Vertical Coverage and Tilt

We generally want to range of audience experiences to be within 6 dB, as much as possible. Putting speakers vertically exactly on-axis can make that really difficult if we don't have a very long spacec, because the front audience member very close to the speaker and like 6 audience members back might have a -12 dB difference. There's two things we can do to fix that: first, we can move vertically instead of horizontally. Then, the distance from the nearest audience member and the distance to the farthest audience member is closer to being the same. Similarly, if we have the space, we can try to move it horizontally farther away. That way, the ratio of distance between the closest and farthest audience member is only 1:2 instead of 1:4 or something.

We should try to aim or speakers such that they are pointed at the farthest audience member in the vertical plane. For a raked audience, this usually menas pointing at the top/back row. We want to try to create a situation where gain differences due to axial position and distance compensate for each other.

3.1 Fill Systems

3.1.1 Downfill

3.1.2 Front Fill

If the stage is tall, or downfill speakers would make the imaging very strange (that is, would make a person's voice sound like it's coming from unreasonably high rather than from themselves), we can instead use a front fill speaker or system. Front fill speakers are placed directly in front of the audience, with their axis pointed at the highest relavent audience memeber out of coverage from the main speaker. In a standard theatre, front fill speakers can only cover about the first three rows. We need to take the same frequency response differences into consideration as with downfill speakers.

3.1.3 Delay

What if we're not out of axial coverage, but out of distance coverage? We can extend the coverage of the main loudspeaker by adding another loudspeaker where we lose coverage (at the 6 dB point). We want in this case to aim not necessarily for the objectively highest audience member, but the highest audience member that we're aiming for with that speaker. This additional distance fill speaker is called a delay speaker (or a delay system, if we have multiple). We call it a delay speaker becase we'll have to delay the signal going to it to account for how slowly sound moves through air.
There are some holes in frequency response, so we will have to include both high and low frequencies.

3.1.4 A Rule of Thumb for Number of Speakers

We can figure out roughly how many speakers we need with Bob McCarthy's formula

$$\frac{\text{Distance to the farthest person}}{\text{Distance to the closest person}} - 1$$

This also tracks if we think in terms of decibels: for example, if this formula gives us 1 because of a $2:1$ ratio, that means there's a -6dB difference from the closest to the nearest, so we only need one speaker.

3.1.5 Output Delay

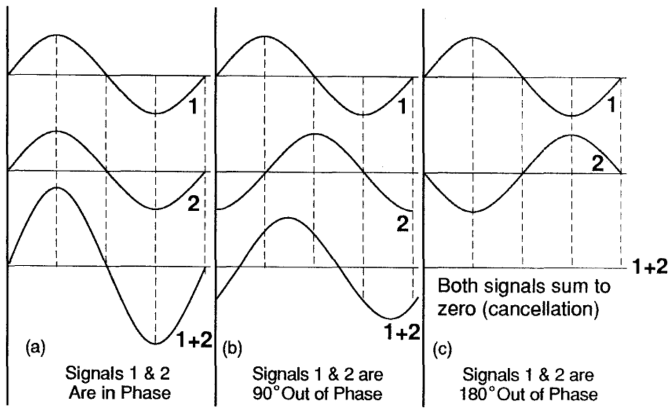
Sound is unique from stuff like light because it's really slow. Light, for example, moves at about $3,000,000$ m/s, whereas sound only travels at 340.29 m/s, or roughly $1,000$ ft/s, or 1 ft/ms. In the time that it would take a beam of light to go around the Earth, a sound would only go about 150 feet.
Especially when we're talking about delay speakers or systems, we need to pay attention to this: we need to add delay equal to the time it takes for the sound from the main speaker to reach the delay speaker.

4 Comb-Filtering: Coverage in the Horizontal Plane

4.1 Phasing/Comb-Filtering

When you hear the same sound from 2 different loudspeakers, or you pick up the sound from two different microphones, it sounds bad in unpredictable ways. As much as possible, we want our sounds to be heard from only one loudspeaker, and we want our sounds to be picked up by only one microphone. We'll be mostly talking about loudspeakers right now.

When two sine waves interact, the resultant combined sine wave is equal to the sume of the two sine waves. We can see how this happens below: if the waves come in at the same time, they'll boost each other by double in every place. If they're coming in 180° out of phase, which is half a cycle behind, they'll cancel each other out exacty. Phase is related to the time that the two waves come in relative to each other (that is, 90° out of phase, means that the second wave comes in 1/4 of a cycle later than the first one).



When two sounds are added, we get interference. There are two types of interference:

(i) Constructuve interference: the two waves are combining so that they're adding to each other—the resultant wave amplitude in that location is greater than either of the waves individually

(ii) Destructive interference: the two waves are combining so that they're subtracting from each other—the resultant wave amplitude in that location is smaller than one or both of the waves individually

Maximum constructive/destructive interference happens when the waves double each other or totally cancel each other out (respectively).

We can also apply this to pink noise: when we delay one pink noise signal compared to another, it forces the resultant pink noise to have nodes (places where certain frequencies are decreased). The farther apart we set the pink noises, the more nodes we have and the lower they go, since lower frequencies require a farther distance to be totally out of phase. This *also* also applies to sounds in general: voices, music, etc. At around 30-40ms later, we hear it as a delay rather than a drequency response issue.

For loudspeakers, the "problem path" is where there's little difference in volume, but noticeable time offsets, because this is what causes more intense comb filtering. We prefer level offsets much more over time offsets, and level offsets can help compensate for time offsets a little bit. This means we want to minimize the region of overlap, because what's where we run into issues of comb filtering (and echo, potentially).

With microphones, we talk about the 3:1 rule: however far your first microphone is from a source, your second source should be at least 3x that distance.

4.2 Focus in Plan: Pan

True stereo isn't really possible for more than a couple poeople at a time down the center line of the speakers. It also gets less exciting the farther back you go—very far back, the angular width of stereo differences gets pretty small, and it's not really that exciting any more. For the people outside of this range, a change in pan just winds up being a level change.

Bob McCarthy gives a rule, which we can use as a starting point but doesn't always work, for how to place loudspeakers. He says to divide the audience in two, then point the loudspeakers at the center of each of these halves.

Important: in the horizontal plane especially, parts of your system *will* overlap in coverage. The important thing to consider is the relationship between level and arrival time: they should either be as similar as possible, or as different as possible.

4.3 Fill Systems in Plan: Extending Horizontal Coverage

Like in the vertical plane, we figure out how to point the speakers for maximum coverage (and minimum overlap), and add fill speakers where there is no coverage to fill in the difference. Like with vertical fill systems, horizontal fill systems are named pretty self-explanatorily, like center fill, side fill, etc.

Part II

Mix Engineering

5 Analog Mix Engineering

5.1 Bus Architecture

A bus refers to console outputs. Buses are organizational tools that collect audio inputs on route to a common destination. For example, if we want to send a mix to the stage manager that's separate from the main mix, we could say that those mixes are on two different buses. Buses are used in analog consoles, digital consoles, recording studios, live sound, DAWs, etc. Although this is in the analog console section, it's not specific to analog consoles, this is just the first time it was convenient to bring it up.

There are a few basic types of busses, which we'll go over in more detail in the following sections:

- Aux (or variable send)
- Group (or fixed send)
- Stereo/mono (or master)
- Matrix

5.1.1 Aux Bus

Aux busses, also known as mix busses, are variable sends—this means that you can control how much sound gets sent to this bus individually, as a function of the fader and the aux send. Aux busses can only receive input channels. Aux busses are usually used for effects sends (eg. reverb) or monitors.

5.1.2 Group Bus

Group busses are fixed sends—this means that you can't independently control how much goes to the group bus, only whether or not sound is going to it. The volume of the channel going to the group is just set by the fader. Group busses can also only receive input channels. Group busses are typically used to group channels together that are headed towards other output busses. A good example is grouping up all the drum mics before sending them to the main output.

5.1.3 Master Bus

Also fixed sends, and also controlled by the main fader. Master busses can receive any combination of input channels and output busses, unlike the last two. When the term “stereo bus” is used to refer to these, it's ued in a generic sense of the term (ie. two speakers), not the literal definition of stereo.

5.1.4 Matrix

Matrices are variable send busses, but matrices can receive only some combintion of output busses. The levels in the sends are a function of matrix sends and the fader level. Matrices are basically aux busses, but for other busses rather than input channels. These are typically used to combine output busses to fill systems.

5.1.5 Pre-Fade vs. Post-Fade

This is only relavent for aux sends and matrix sends. These sends can be pre-fade or post-fade. Pre-fade busses mean that the audio is diverted to the aux bus before it's sent to the fader, so both the fader and the send get the same input (ie. the send volume doesn't rely on the fader—the main mix will have no effect on the bus). Post-fade busses have audio diverted after the fader, so the input for the send is the same as the output of the fader (ie. the send volume also relies on the fader—changing the main mix will change the volume going to the bus as well).

6 Additional Considerations for the Engineer

6.1 RF Basics

RF stands for radio frequency—it's the umbrella term for all wireless devices. We're familiar with air traveling through air/water/etc as acoustical signal (ie. pressure waves). We're also familiar with the idea of audio travelling through cables as voltage information. RF is similar—the audio signal moves through the air as electromagnetic signal (like light), which can be decoded by the receiver.

Like audio, the electromagnetic waves that the signal travels on has amplitude and wavelength/frequency. Note that these refer to the characteristics of the transmission, not the characteristics of the sounds they carry. Unlike sound waves, RF waves travel at the speed of light, and have some polarization (ie. the waves are aligned in some axis)—this means that antennae have to be in the same direction of the polarized signal.

RF transmission is referred to as TX, and receiving is referred to as RX.

Receivers work by taking a small band of frequencies in the EM spectrum and listening to them. This is usually written as a single frequency, but in reality, it's a narrow region around that frequency. Receivers have no way of knowing whether what they're picking up is what it's supposed to be, or just unrelated noise at that frequency. Our job is to make sure that the RF signal is the strongest thing at that frequency so we don't have to worry about this.

Some types of attenas:

- Whip antenna
  Most basic kinds, omnidirectional. Comes in half-wave and quarter-wave sizes. Quarter-waves are less precise, although tbh neither are particularly precise. They're fine if you don't have a lot of noise, but not very preferable otherwise.
- Log periodice (paddle, sharkfin)
  Cardioid directional pattern, roughly 150° wide.
- Helical
  Thinner conic pattern (60°), picks up any polarization.

RF distro (distribution) allows one pair of antennas to feed multiple receivers.

Bodies absorb RF really easily. We want to put our receivers high in the air, giving it a good line of sight. In general, if an obstacle is small compared to the wavelength of the signal, it won't affect it much, but if the obstacle is large, it will block it much more. The inverse is true for windows or holes.

We also sometimes need to worry about multi-path interference: that is, sometimes RF signal will reflect off of a surface, and antennas will pick up both the original signal and the reflected signal. This is a lot like comb filtering: the two signals are arriving out of phase, which can mean we have huge attenuation of signal. This is why we can randomly lose signals in certain places. In order to fix this, we worry about diversity: we keep multiple (usually 2) antennas at one time, and the receiver will use the signal from whichever antenna is getting the strongest signal. In order to achieve multi-path diversity, we actually want to keep them pretty close to each other (they should be within 1/4 to 1 wavelength apart). In our RF operating ranges, this means 3-12 inches for high frequencies and 12-46 inches for low frequencies. We can usually call this "about a rack width apart."

There are other kinds of diversity we might want to pay attention to as well:

- Coverage area
  You can have multiple antennas in two different areas, of one of the antennas can't cover the whole area. THere will still be multi-path dropout.
- Polarization
  You can have one antenna vertically and one horizontally, so that if signal polarization changes, we can still get signal. We can combine this with multipath diversity by having both at 45 degree angles.
- Obstacles
  Similar to coverage area, if there's going to be a big obstacle, we can use an antenna to compensate.

One last form of interference: intermods/intermodulation interference. Every time we introduce a new transmitter, it introduces some ambient noise and random peaks from interacting with the other frequencies, mucking up the RF spectrum. The locations of these depends on the relationship between the frequencies. The height depends on the distance between the transmitters. The main solution is to calculate the intermods when selecting the operating frequencies to make sure that we won't be getting intermods where we have other devices operating. Fortunately, there's a lot of software that can calculate this for you.

6.2 Analog Patching

Patching refers to plugging the output of one device into the input of another. It refers to the signal flow between devices. This is not to be confused with routing, which lays out the signal flow within a single device. In a sound system, things don't need to be hard-patched to each other in an inflexible way. We can include patchbays at various points in the signal flow, allowing you to flexibly patch the flow where you want it. This means that we can just plug inputs/outputs where it's most convenient and patch them where they need to be, rather than needing to run it all the way to the right channel when it might be more difficult that way.

Some patchbays operate in “normal” configuration. This means that there’s a default when no cables are plugged into the front, where the inputs and outputs are correlated 1:1, and plugging in a patch cable overrides that internal connection.

7 Stages of Amplification

7.1 Gain Structure

Throughout the audio signal chain, there are a lot of places to adjust gain (ie. the strength of the electrical signal which correlates with volume). Where and when do we want to be adding or removing gain? There are 4 big places to change gain: preamp, volume fader, output fader, and amplifier. When you bring up the signal too high at the early stage, it's easier to get clipping, and if it's too low at an early stage, you wind up having to amplify later in the signal chain, picking up all the extra noise in the system. Clipping can happen at any point in the gain stage, but sometimes for different reasons (ie. if the amp gain is too high and causes clipping, it's usually because the speakers themselves are being driven too hard and might be damaged, but at the preamp level, you're just trying to push through more voltage than it's able to). You want to be up as loud as possible without clipping early in the signal chain (leave yourself some headroom).

7.2 Power Amplifiers

Power amps (usually just called amps) come right before the speakers. Amps convert line level signal (1-10 V) to speaker level signal (10-50 V). Amps are not transiducers because they don't convert between types of energy, they just modify the electrical signal. Ideally, the amps don't do anything other than make the sound signal louder (although they will always color the sound in some way, like all components in the signal chain).

Stereo mode: each input goes to its respective output (1 to 1, 2 to 2). Bridged mono: the first input goes to both outputs, the second input is ignored. Bridged mono is useful because it can produce more power than stereo mode, which is helpful for bigger speakers.

We can daisy-chain loudspeakers together: we can send the same signal to each speaker in a chain by physically connecting them together. This assumes, of course, that we have enough power coming from the speakers to do this for the number of speakers we're trying to chain together. This will decrease the impedance of the load by putting the speakers in parallel (ie. two 8 Ω speakers will have a total impedance of 4 Ω). Usually, if we use more than 3 loudspeakers on a single chain, we start getting into fire hazard territory, so we don't want to go above that. The amp spec sheet will tell us the power the amp can deliver to different impedances. If the amp doesn't give a spec for a particular load, don't try it, because that can be a fire hazard (ie. don't go lower than the minimum impedance listed).

We also have to consider how much power we're delivering to a loudspeaker, usually called power handling, power rating, or power capacity (but not always—just look for the measurement in Watts). We want to make sure the amp isn't providing more power than the speaker is able to handle, breaking the speaker. Many list stats for continuous (rms), program, and peak power. Continuous refers to an indefinite sine wave, program refers to a more complicated signal (a shorter speech/music source, for example), and peak refers to an impulse response, like a snare drum hit or a plosive in a speech. When you're planning amps on a circuit, always plan for the max (probably)—if the spec sheet tells you a max current draw, that's also ideal.

In order to determine matching up speakers and amps, we need to first match up the impedance of the loudpspeaker to the impedance ratings of the amps, and the power capacity of the speaker vs the power driven by the amp to a speaker with that impedance. As a rule of thumb, we want the peak power of the amp to be in the ballpark of half of what the speaker can handle (anything more than that runs potential damage to the speaker).

We also want to look at sensitivity, which tells us how loud a speaker is in response to a given amount of power. It will be given as dB SPL 1W @ 1m distance (ie. decibels of sound pressure level from a 1 Watt power delivery at 1 meter distance). Higher sensitivity speakers will be louder, lower sensitivity will be quieter at a given power. We can easily calculate the level at other distances using the inverse square law (-6 dB per doubling). We can also calculate the level for other amounts of power by calculating the ratio of power delivered against 1 W.

8 Paperwork

There are four pieces of paperwork we should be responsible for as systems designers. The first two are our plan & section plots like we've already talked about, the other two are our patch list, and our block diagram/signal flow diagram/system overview. These last two basically show the same information, just in different forms (one in writing, one graphically).

The paperwork is an opportunity to share with your collaborators anything they'll need to know—make sure to include everything they need to know (and nothing they don't). The paperwork we're making here is largely used for communication between the designer and the engineer, but, for example, the plots will be used by other designers, technical directors, stage managers, etc.

Audio paperwork is really poorly-standardized. There have been occassional attempts to standardize it (most recently in 2008 by USITT for block diagrams, the documentation for which is also in this folder), but most have been pretty unsuccessful. Because of that, we should consider on a case-by-case basis what we should include/exclude/label based on our audience rather than just following some rote guidelines.

In our plots, we can (and usually should) include annotations–telling us, for example, what kind of speaker we're looking at, how many there are if they're stacked in that view, etc. In something like vectorworks, we use the Callout tool on the sheet layer (NOT the design layer). In our annotation, include: speaker number, speaker function, make/model, rigging.

The patchlist shows detailed written information showing the signal flow between all the elements of the system (usually written in a spreadsheet). The block diagram, similarly, is where we bring the patchbay visually to life. Usually, each element is described with a labelled block, and every connected item is connected with a line.

9 Microphones

9.1 How Microphones Work

Microphones are transducers: electronic components that comvert one type of energy into another (in this case, from acoustical energy to electronic energy, the opposite of speakers). There are three basic families of microphones:

(i) Condenser
Condenser microphones consist of a small parallel-place capacitors (that is, two flat parallel conductors). The distance between the plates affects the voltage across the plates. Acoustical energy moves one of the plates, causing a change in voltage proportional to the vibrations of the acoustical energy. In order for this to work, though, we need some charge on the capacitors (which we do with phantom power, or sometimes just a battery). The diaphram of a condensor microphone can be very light, making them potentially much more sensitive (but also less durable).

(ii) Dynamic
Dynamic microphones, also called moving coil microphones, are the most basic kinds of microphones. The vibrations in air move a coil of wire around a magnet, which generates current proportional to the sound vibrations.

(iii) Ribbon
Ribbon microphones are the most complicated. In these, we have a magnet, and suspended in the middle of the magnet is a very thin strip of corrugated metal (the ribbon). Acoustic energy moves the ribbon in the magnetic field, generating a current, much like dynamic microphones.

9.2 Selection and Deployment of Microphones

9.3 Feedback

9.4 A:D and D:A Conversion

10 Digital Mix Engineering

10.1 DCA Assignments

10.2 Digital Consoles

11 Delaying Larger Systems

11.1 Input Delays

11.2 Delay Matrixing

11.3 Staging Zones

12 System Principles from Concert Audio

12.1 Subwoofer Deployment

12.2 Cardioid Subwoofers

12.3 Line Array Theory

13 Control Systems and Networked Audio

13.1 MIDI, OSC

13.2 Networking Basics

13.3 Dante Overview

14 System Tuning and Measurement Microphones

14.1 SMAART Software and Approach