PDHonline Course E126 (2 PDH)

Sound System Design for Cafeterias, Auditoriums and Small Churches

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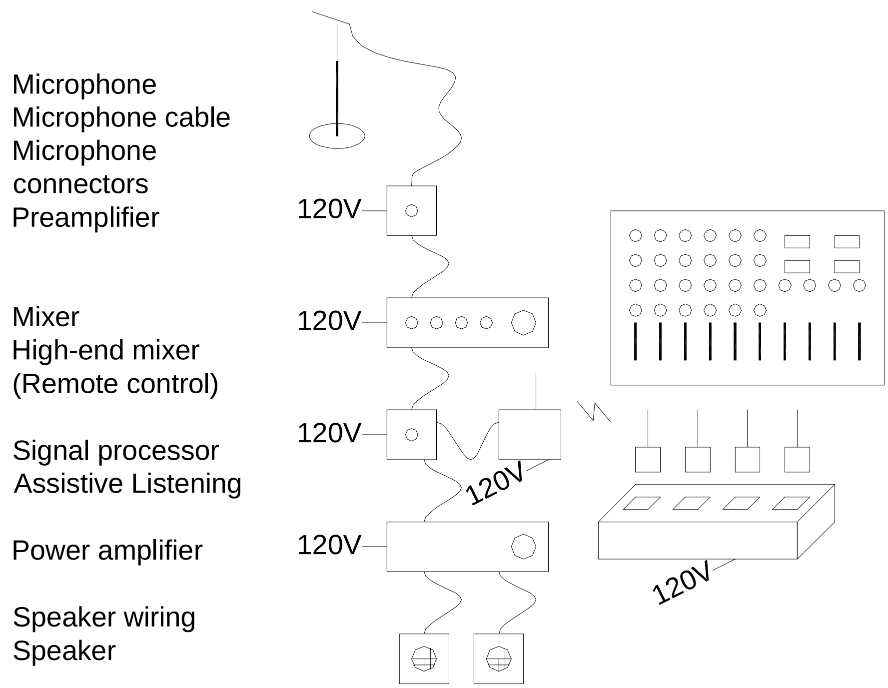
Course Content

Sound systems have been in use since the 1940's, when Frank Sinatra first used a microphone to convey intimate music to the audience. Early sound systems were limited by tube-based electronics - noise, frequency limits and non-linearity. Modern solid-state electronics have almost unmeasurable noise, response well beyond human hearing and linearity beyond human detection. The remaining problems are selection of equipment, location and installation of the equipment and room acoustics. These problems, some underlying definitions and examples are illustrated and addressed after the system graphic below:



Sound System Graphic

- Sound characteristics - waves, complex waves, frequency, low frequency, high frequency, linearity, noise, sound intensity.
- Room acoustics - size, reflectance, resonance, aesthetics.
- Sound system components - sound lectern, microphone, microphone cable and connectors, preamplifier, mixer, high-end mixer, signal processor, power amplifier, speaker wiring and speaker.
- Sound lectern - power source, components, limitations, cost.

- Microphone - types, power source, pickup pattern, limitations, cost.
- Microphone cable - conductors, shield, jacket, limitations, cost .
- Microphone connectors - types, conductors, shield, limitations, cost.
- Preamplifier - types, limitations, cost.
- Mixer - types, limitations, cost.
- High-end mixer - features, limitations, cost.
- Signal processor - functions available, limitations, cost.
- Power amplifier - features, limitations, cost.
- Speaker wiring - conductors, jacket, limitations, cost.
- Speaker - types, limitations, cost.

Each of these concepts and components will be discussed in some detail.

Sound Characteristics:

Noise is the phenomenon of vibrations reaching your eardrum and being transferred to your brain where no meaning or pattern is detected. Sound is the same thing, except your brain detects music or a voice message or a meaningful signal tone.

Vibrations which can be converted by your eardrum are in the range of 20-15,000 Hz with an intensity of 0-100 dB(C) SPL.

"Hertz" is the new name for "cycles per second", The way you find the frequency, or Hertz, of a vibration is by connecting it to a spectrum analyzer. Until recently, a spectrum analyzer was a sophisticated $10,000 instrument. Today, it is a free software program you can download from the internet. (See *CoolEdit* in the references at the end of the course.)

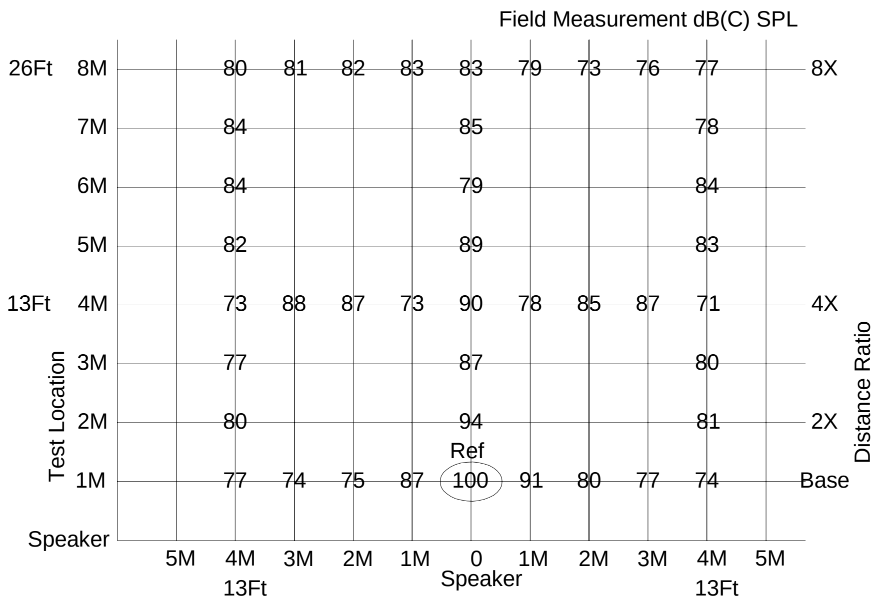
The display on a spectrum analyzer is frequency on the horizontal axis and amplitude on the vertical. You can adjust the controls for wide-spectrum, 20-15,000, or your choice of narrow spectrum, perhaps 300-7000, the critical voice frequencies. The spectrum gives graphic evidence of low-end roll-off (the lowest tones present) and high-end roll-off (the highest tones present). Over time, it shows how uniform the amplitude is over the range and if any peak frequencies are present. When no signal is present, it shows the amplitude and frequency of system noise.

With very little trouble, the spectrum analyzer produces numeric measurements of the actual spectrum present, linearity, peaks and noise. This is how to scientifically compare music reproduction systems.

To test the air-borne music quality and room acoustics, you need a microphone which has better response than the system being tested. A studio-quality large-diaphram condenser microphone costs about $60 and plugs directly into the computer sound card.

It is interesting to watch the spectrum of an Elvis Presley song, but for sound system design, pure tones and pink noise are used as sources. Both of these sources are available as free software programs from the internet. (See *SweepGen222* in the references at the end of the course.) A single pure tone is useful for multiple measurements of the system. The speaker distribution chart below was made at 1,000Hz, a common test frequency. Obviously, the results relate only to system performance at 1,000Hz. The non-uniformities in the sound pressure level gradient and the extreme sensitivity to sound meter location are results of interferences produced by a pure tone in a real-world space. Fortunately pure tones rarely exist outside the laboratory.

The goal of the measurements was to verify the predicted drop-off in sound level and determine the off-axis performance of a commercial enclosed speaker.



Free-Field Sound Distribution, 1,000Hz
20W Bass Guitar Amplifier(System)

Field Note: Very location sensitive measurements.
1/2-in meter location shift produces +/- 3dB value shift.

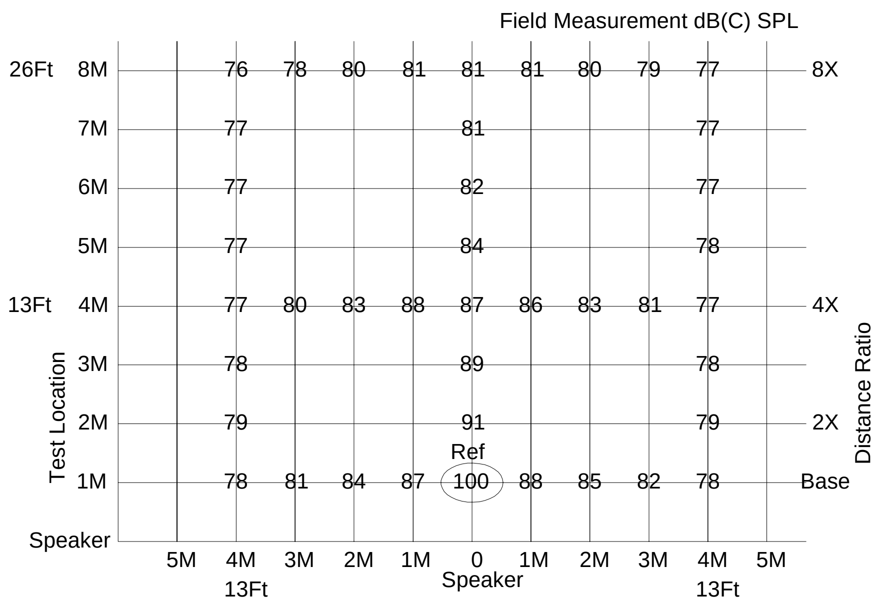
On-Axis Data Analysis

Location	Distance Ratio	Measured dB(SPL)	Theory dB(SPL)	Difference	Percent
1M (3Ft)	Base	100	100	0 dB	
2M (6Ft)	2X	94	94	0 dB	
4M (13Ft)	4X	90	88	2 dB	-2.3%
8M (26Ft)	8X	83	82	1 dB	-1.2%

A more valuable pure tone test uses the sweep generator function. You can adjust the start and end frequencies and the rate of “sweeping” from the lowest to the highest. A sweep generator is very good for manual tests. For instance, you can check the low-end roll-off and high-end roll-off of a sub-woofer system by connecting the sweep generator and listening. No measure of linearity, but clear measure of system range. Because of multiple frequencies, interferences effects cancel out.

A sweep generator into a system and a recorder on the output produces a record of system performance which can produce many interpretations in off-line analysis. (Remember the requirement for a microphone which is better than the system being tested and expected interferences.) The off-line analysis can be very tedious and technical, but produces heavy-duty results.

A quick, graphic way to evaluate overall system performance is to put in “pink noise” and read the output on the spectrum analyzer. “Pink noise” is a random collection of all frequencies in the music range, 20-15,000Hz with uniform amplitude. A very consistent signal goes in; any non-linearities are measures of the system under test and immediately visible. The signal can be recorded or a screen-capture can archive the results. (Remember the requirement for a microphone that is better than the system being tested. Interferences are valid measures, so multiple microphone locations are essential - even 12-in apart.)



Free-Field Sound Distribution, Pink Noise
20W Bass Guitar Amplifier(System)

On-Axis Data Analysis

Location	Distance Ratio	Measured dB(SPL)	Theory dB(SPL)	Difference	Percent
1M (3Ft)	Base	100	100	0 dB	
2M (6Ft)	2X	91	94	3 dB	-3.3%
4M (13Ft)	4X	87	88	1 dB	-1.1%
8M (26Ft)	8X	81	82	1 dB	-1.2%

“Free field” means that no walls or objects interfere with the sound around the source and test instrument. For valid, scientific testing, an anechoic chamber is used. This is a room with absorbent walls. For the data collected above, the author’s front yard was used, with the source about 8-ft above the ground, sitting on the edge of an SUV and the author walking around a grid on the grass, holding the sound meter above his head. As with all field data, the discrepancies from theory are as meaningful as the demonstration of the theory. Note, however, the uniformity on-axis and off-axis of the low-cost commercial sound reinforcement system.

The measurement of sound, dB(C) SPL, confuses many. For our purposes, it is a valid numeric measure of sound intensity that closely matches human ear response. The human ear/brain is logarithmic in response. That is, increases of 2x, 4x, 8x and 16x sound like equal increments. In dB notation, these are 3dB, 6dB, 9dB and 12dB. Historically, 1dB is the smallest change perceivable by the human ear. SPL means Sound Pressure Level, indicating that the measurement started with a microphone. (C) means linearly-weighted. The A-weighting is limited to voice frequencies. The 100-ref means that the system under test was adjusted to 100 dB(C) SPL at a point 1M in front of the speaker, then the measurements were made. 100 dB SPL at 1M is a very standard speaker output for 1 watt of power being applied to the terminals.

To provide an intuitive understanding of sound pressure level, the residential neighborhood background noise for the free-field test was less than 50 dB (the lower limit of the meter used); 80 db is the noise level of a new Toyota Corolla traveling at 70 mph, 85 dB is the preferred radio sound level in the Corolla at 70 mph, and 85dB is the OSHA limit for 8-hr exposure in the workplace. 100db is generally considered to be the pain threshold (which was confirmed during the tests).

The use of dB is widespread in sound system specifications. A 10-watt amplifier is 10dB stronger than a 1-watt amplifier. A 100-watt amplifier is 20dB stronger than a 1-watt amplifier. The 100-watt amplifier (operating at full rated gain) amplifies the 80 dB Toyota noise to 100 dB (pain). This is why it is important to record in a very quiet studio or edit out all no-music or no-voice portions.

dB	Power Ratio	Voltage or Current Ratio	dB	Power Ratio	Voltage or Current Ratio
0.0	1.00	1.00	10	10.0	3.2
0.5	1.12	1.08	15	31.6	5.6
1.0	1.26	1.12	20	100.0	10.0
1.5	1.41	1.19	25	316.0	18.0
2.0	1.58	1.26	30	1,000.0	32.0
3.0	2.0	1.41	40	10,000.0	100
4.0	2.51	1.58	50	10exp5	316.0
5.0	3.16	1.78	60	10exp6	1,000.0
6.0	3.98	2.00	70	10exp7	2,162.0
7.0	5.01	2.24	80	10exp8	10,000.0
8.0	6.31	2.51	90	10exp9	31,620.0
9.0	7.94	2.82	100	10exp10	10exp6

dB(power) = 10log(P2 / P1)
dB(voltage) = 20log(V2 / V1)
dB(current) = 20log(I2 / I1)
dB(SPL) = 10log(SPL2 / SPL1)

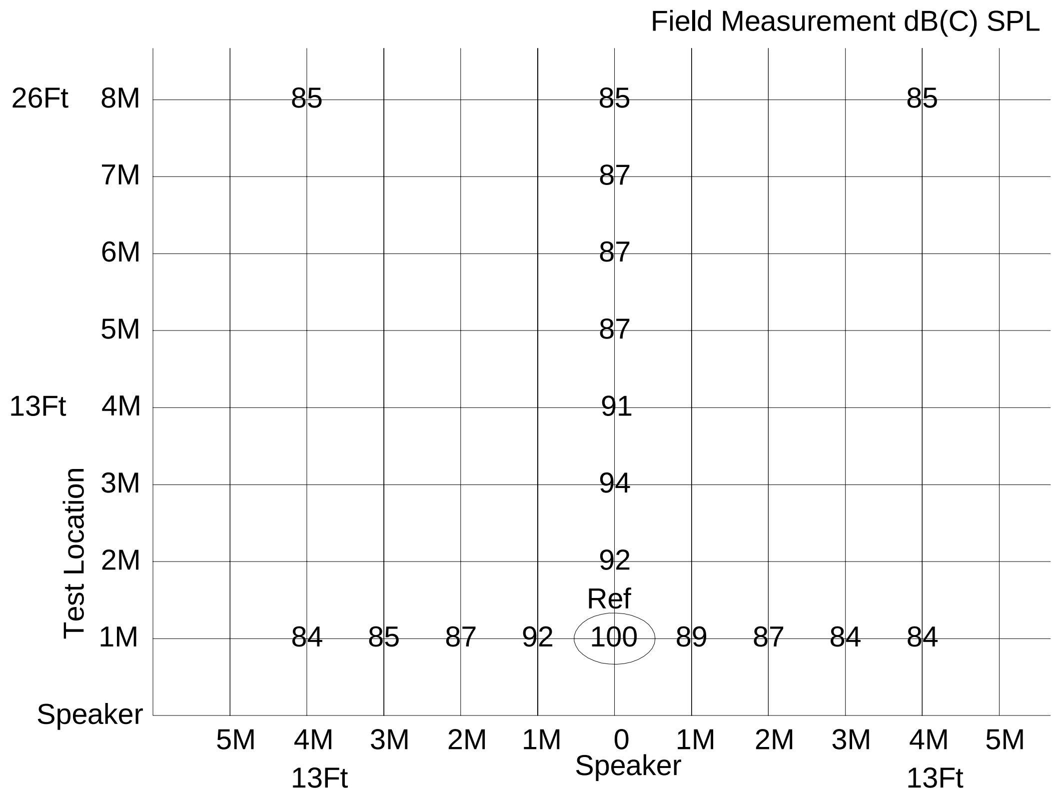
Room Acoustics: .
There are two problems with room acoustics - reflections and resonances. Reflections are largely undistorted echoes. In small rooms, the echoes come back while the word is being spoken and cause no difficulty beyond limiting the gain available before feedback occurs. In a large room, legitimate theater, music hall or cathedral, the reflections seriously interfere with intelligibility.

Resonances are non-uniform reflectances. There is always one frequency that reflects more than the rest, sometimes very much more than the rest. This is the tone that first causes feedback squeal.

Reflections can be measured with a pulse test that is not pursued here. Resonances are easily recognized by spectrum analysis of system response with a pink noise source. It is valuable to know the resonant peaks, because they can be diminished by signal processing in the sound system, permitting larger overall gain and more pleasant listening. Some purists object to response adjustment to eliminate feedback, but remember that the room acoustics are providing very non-linear response before the adjustment.

Qualitatively, a space with many reflections and resonances is referred to as “live” and is easily recognized by hard surfaces. A space with few reflections and resonances is referred to as “dead” and is recognized by soft surfaces, carpeting, upholstered furniture, possibly tapestries on the walls.

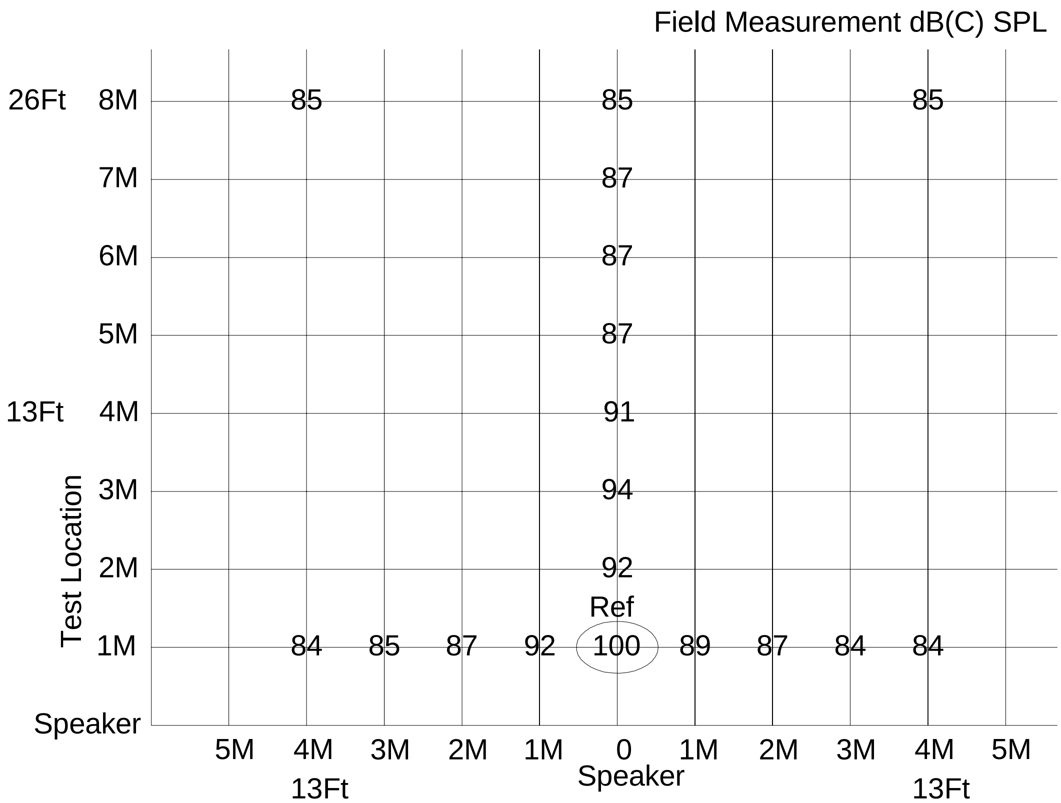
The same tests were performed in a conference room used for employee informational presentations and web casts to remote offices. The goal was to see what effect the reflecting walls have on sound levels.



"Live" Conference Room, Pink Noise
20W Bass Guitar Amplifier(System)

On-Axis Data Analysis

Location	Distance Ratio	Measured dB(SPL)	Theory dB(SPL)	Difference	Percent
1M (3Ft)	Base	100	100	0 dB	
2M (6Ft)	2X	95	94	1 dB	+1.1%
4M (13Ft)	4X	93	88	5 dB	+5.7%
8M (26Ft)	8X	92	82	10 dB	+12.2%



"Live" Conference Room, Pink Noise
20W Bass Guitar Amplifier(System)

On-Axis Data Analysis

Location	Distance Ratio	Measured dB(SPL)	Theory dB(SPL)	Difference	Percent
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8M (26Ft)	8X	85	82	3 dB	+3.7%

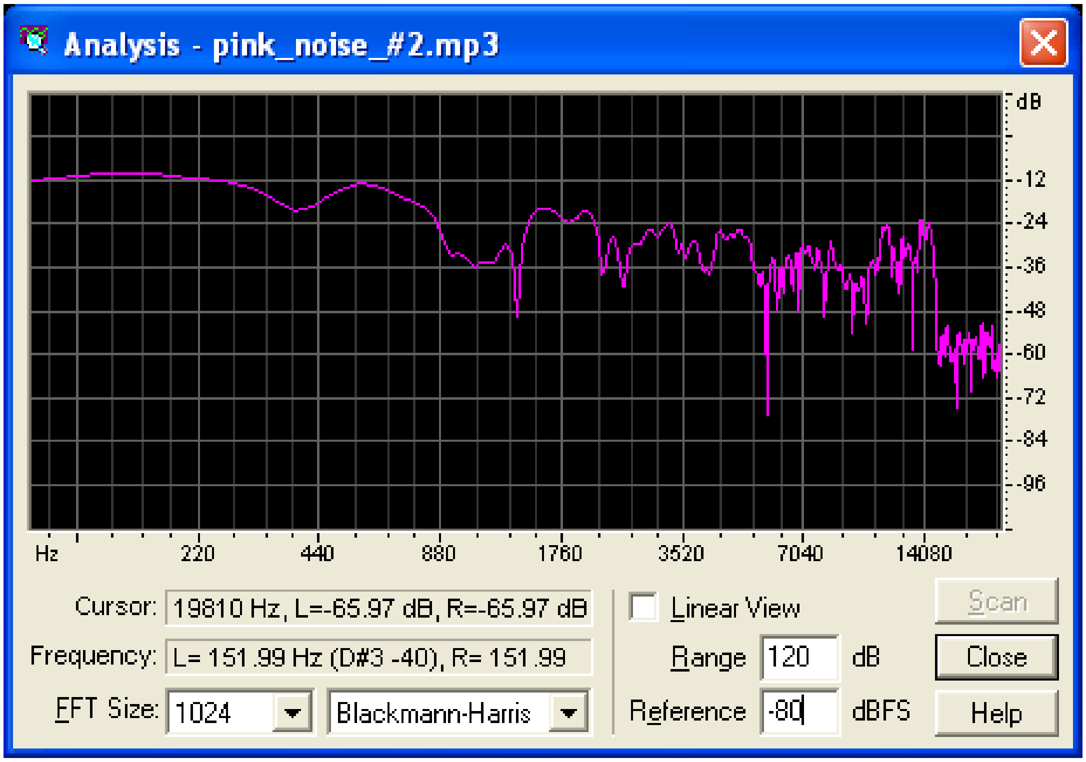
1950's Sound Design Textbook Recommendations for Many Small Speakers in Ceiling

(Standard 30-ft x 30-ft classroom or conference room = 900 sq-ft.)

Example *	Ceiling Height .*	Ceiling Height -4 ft	Area per 60-deg spkr, sq-ft	Speakers per 1000 sq-ft	Mounting centers, c-c, ft. *	Required Watts per 1000 sq-ft*	Area per 90-deg spkr	Speakers per 1000 sq-ft	Mounting centers, c-c, ft *	Required Watts per 1000 sq-ft*
	7.0	3.0	9.0	166.0	2.5		28.0	52.0	4.4	
	8.0	4.0	16.5	91.0	3.3		50.0	30.0	5.8	
	9.0	5.0	28.5	52.0	4.4		78.0	20.0	7.1	
classroom, office	10.0	6.0	38.5	39.0	5.1	40.0	114.0	13.0	8.8	13.0
	11.0	7.0	50.0	30.0	5.8		153.0	10.0	10.0	
	12.0	8.0	65.5	23.0	6.6		200.0	8.0	11.2	
	13.0	9.0	85.0	18.0	7.5		256.0	6.0	12.9	
	14.0	10.0	106.0	14.0	8.5		314.0	5.0	14.1	
cafeteria	15.0	11.0	128.0	11.0	9.5	50.0	380.0	4.0	15.8	25.0
	16.0	12.0	154.0	10.0	10.0		452.0	4.0	15.8	
	17.0	13.0	176.0	9.0	10.5		530.0	3.0	18.3	
	18.0	14.0	205.0	7.0	12.0		620.0	3.0	18.3	
	19.0	15.0	235.0	6.0	12.9		710.0	2.0	22.4	
	20.0	16.0	271.0	6.0	12.9		810.0	2.0	22.4	
	21.0	17.0	308.0	5.0	14.1		910.0	2.0	22.4	
party center	22.0	18.0	345.0	4.0	15.8	100.0	1020.0	2.0	22.4	50.0

* = column added to original table

The spectrum of the pink noise source used is reproduced below. This display indicates very good representation of frequencies between 100 Hz and 14 kHz, with a missing frequency at 5600 Hz and 24dB drop-off at 14 kHz.



As with all electronic and acoustic measurements, the limitations of the test sources and test measures must be well below the values of the variables being measured. The system is almost always limited by the microphone and speaker selected, then by the space acoustics. Very good microphones are available in the $50-100 range and usually identified as "music, wide range" or "voice, low-frequency roll-off".

Excellent speakers are available in the $100-400 range and can be identified for application by which elements are inside the enclosure. No tweeter, no high frequencies. General-purpose speakers all have mid-range speakers for voice. A woofer of 10" or 12" is required for low frequencies to about 150 Hz. A sub-woofer is required to go down to about 40 Hz. One major manufacturer advertises small speakers with low frequency response. The speaker has very poor low frequency response but must be used with the manufacturer's equalization system that overdrives the small woofer to get sound levels. This fact is hidden in a footnote on page three of the specification. The quoted response is with the equalization. They will not reveal the direct-connect response.

Room acoustics are complicated, with limited range of ameliorative actions available. The reason for this course being limited to cafeterias, auditoriums and small churches is that these spaces is to avoid addressing the acoustics of large, complicated spaces. Small spaces respond well to a number of speakers in the ceiling or around the perimeter. A microphone with built-in low end roll-off eliminates room noise.

A last limitation of acoustic tests is digital recording of acoustic data for off-line analysis, as with the spectrum analyzer. Microsoft .WAV files do not use compression, and record with the accuracy of your equipment. Generic .MP3 codecs have adjustments that limit the accuracy of the recording in order to reduce file size. Such limitations are at the limit of the audio equipment and certainly do affect the observed results. Using an identical test setup assures that artifacts not produced by the system under test are at least identical, if not minimal.

As discussed previously, the purpose of pink noise is to eliminate acoustic or electronic effects that operate on narrow frequency bands. By having all available frequency bands (as does music) the system "average" performance is tested.

Sound System Components:
The Sound System Graphic in the Introduction portion of the course indicates the components which must be considered in designing a sound system. In a small sound system, several components may share a common enclosure and carry a slightly different name.

Two completely self-contained sound systems are not being discussed here - the bullhorn and portable PA system. The bullhorn is a hand-held device favored by protesters and SWAT teams in TV movies.

The portable PA system is a more gentile version that has a short cord on the microphone and is favored by Lolly-the-Trolly tour guides.

Sound Lectern: .
A sound lectern almost exactly matches the bass guitar amplifier used for acoustic testing. It contains a good mid-range speaker, a good amplifier, a good preamplifier and adds a good microphone. The enclosure is superb, usually costing more than the components. Cost is $1,000-4,000. Outboard speakers are available, permitting use in very large spaces. Note that security is excellent, as the unit can be rolled to a closet and locked up.

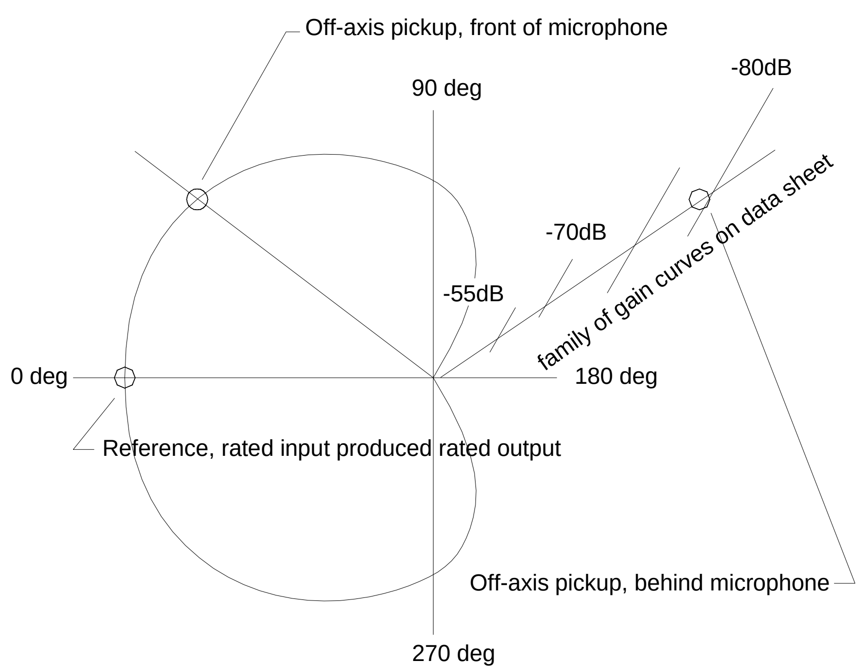
There are aesthetic objections to sound lecterns along with budgetary complications. A sound reinforcement system is considered a capital expenditure. A sound lectern is considered moveable furniture. They have different approval chains and cannot be interchanged.

Microphone: .
Microphone selection evokes considerable emotional response. All but the cheapest $1 mikes that come with tape recorders work well. Some work better in critical situations. A key question, however is directional vs non-directional. Directional is better for a single speaker. Non-directional is better for a group. There is no price or performance difference.

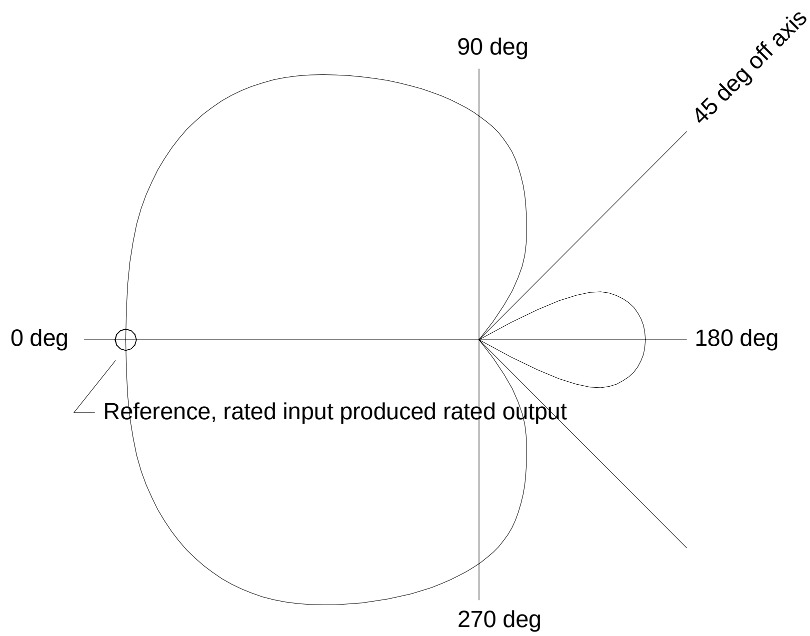
Microphone descriptions identify directionality, sound element, output level and frequency response. There is considerable overlap among types and some imaginative advertising. The author's collection of experience and catalog reading is summarized as follows:

Microphone Summary

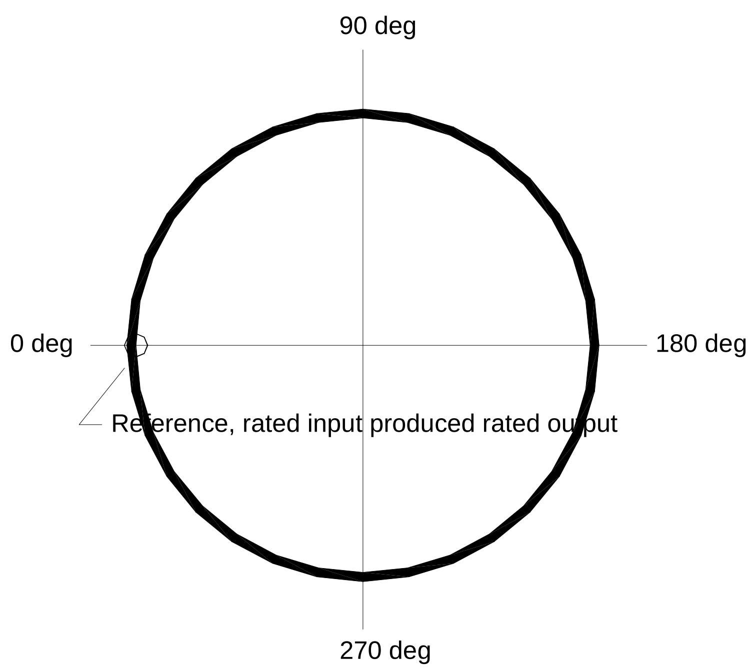
Name	Directionality	Element	Level	Response	Price	Comment
Crystal, hi-z	Omni	Crystal.	-50dB	300-6000Hz	$1-30	Bad
	Cardioid	Crystal	-50dB	300-6000Hz	$1-30	
Electret #1	Omni	electret	-55dB	300-6000	$1-30	No amp, Bad
Dynamic	Omni	dynamic	-70 to – 60dB	100-10,000Hz	$20-200	Wide range of features
	Cardioid	dynamic	-70 to – 60dB	100-10,000Hz	$20-200	Wide range of features
	Super cardioid	dynamic	-70 to – 60dB	100-10,000Hz	$20-200	Wide range of features
Electret #2	Omni	Electret	-60 to –45	50-15,000 Hz	$50-200	Req batteries
	Cardioid	Electret	-60 to –45	50-15,000 Hz	$50-200	Req batteries
Condenser	Omni	Diaphragm	-60 to -50	30-18,000Hz	$60-1000	Req DC pwr
	Cardioid	Diaphragm	-60 to -50	30-18,000Hz	$60-1000	Req DC pwr
Condenser	Omni	Ribbon	-60 to -50	30-18,000Hz	$200-1000	Req DC pwr
	Cardioid	Ribbon	-60 to -50	30-18,000Hz	$200-1000	Req DC pwr



Microphone Cardioid Pickup Pattern



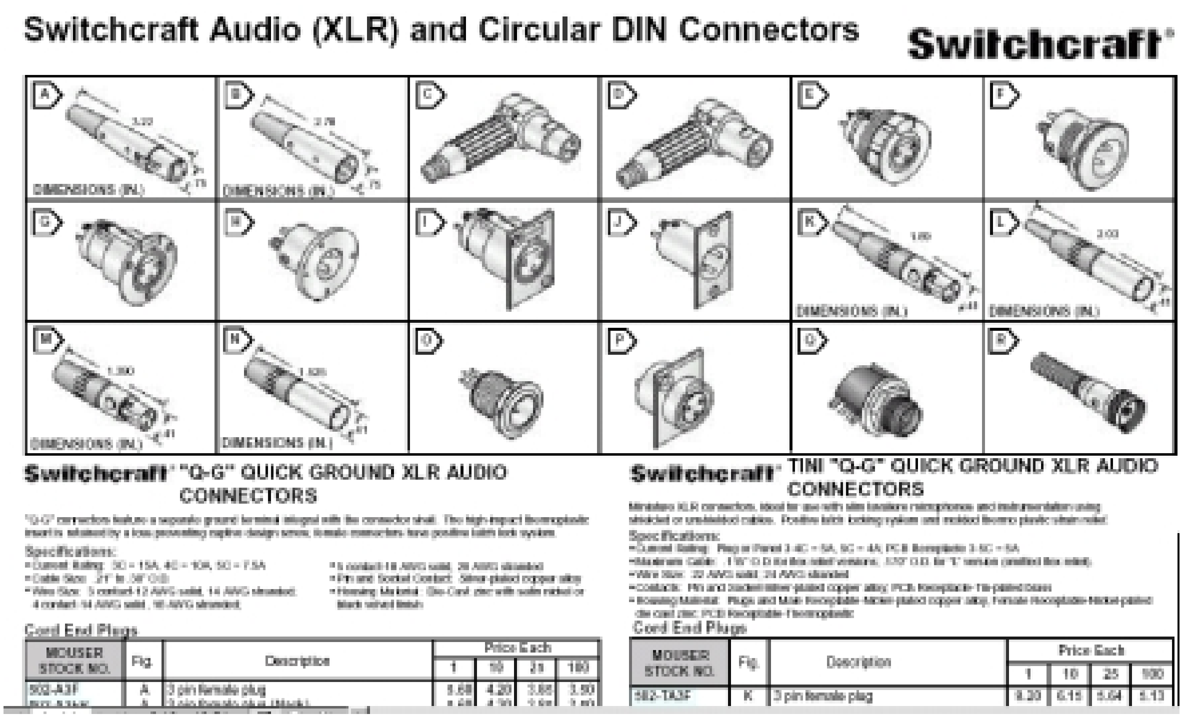
Microphone Super-Cardioid Pickup Pattern



Microphone Omnidirectional (non-directional) Pickup Pattern

It would be valuable to distinguish microphones by survivability. Unfortunately, all rate from low to extremely-fragile. Condenser microphones are sensitive to moisture, in addition to vibration and impact. A spit-screen is desirable for condenser mikes for performers.

Microphone Connectors: .
Three types of microphone connectors must be considered, along with myriad variations. They are XLR-3, TRS, and mini. Many years ago Cannon and Amphenol made XLR connectors for broadcast microphones. Today, Switchcraft makes them for broadcast, commercial sound and a wide range of control applications. Mouser Electronics also lists five new manufacturers. The “3” indicates a source conductor, a return conductor and a shield conductor, connected internally to the shell of the connector. When used for unbalanced inputs, the return and shield are connected to the same pin. XLR-3 connectors are available as male cable connectors, female cable connectors, super-cheap plastic cable connectors, chassis male, chassis female, super-cheap molded chassis connectors and printed circuit mount connectors. The plastic connectors do not have shielding at the terminations and often lack the positive lock on the female piece that grips a groove in the male piece. The following graphic shows the various forms.



TRS are the phone plugs, where phone is short for telephone. Three-conductor ¼-in phone plugs were used on the telephone switchboards of the 1920’s. Per telephone lingo, they are TRS plugs, for Tip, Ring and Shell. This exact plug is used for balanced microphone and high-level audio connections and works extremely well. Tip is supply; ring is return and shell is shield. For two conductor versions, the return and shield are both connected to the shell. As with XLR connectors, a metal shell is shielded; a plastic shell passes noise and hum. Many, many versions of TRS plugs, jacks and adapters are available. A few are listed on the following graphics.

Switchcraft® LITTEL PLUG PHONE PLUGS

Solder terminals include built-in cable clamps. Shielded metal handles are nickel plated brass. Maximum cable diameter is .297" (7.54mm).

A

2-Conductor Illustrated

B

2-Conductor Illustrated

C


2-Conductor Standard 1/4" I.D. Types Typical mating type: 11

* 502-40: .28" cable O.D., 502-70: .38" cable O.D., 502-170: .25" cable O.D.

MOUSER STOCK NO.	Fig.	Handle	Terminal	Price Each		
				1	10	25
502-240	A	Black	Screw	3.40	2.55	2.34
502-250	A	Black	Solder	3.20	2.40	2.20
502-255	A	Red	Solder	3.20	2.40	2.20
502-270	B	Shielded	Screw	4.76	3.57	3.27
502-280	B	Shielded	Solder	3.96	2.97	2.72
502-40	A	Black	Screw	4.40	3.30	3.03
* 502-70	B	Shielded	Screw	6.00	4.50	4.13
* 502-170	C	Shielded	Screw	7.00	5.25	4.81
* 502-220	D	Black	Screw	4.76	3.57	3.27
502-226	E	Shielded	Solder	4.36	3.27	3.00
502-227	D	Black	Solder	3.68	2.76	2.53

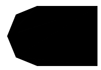
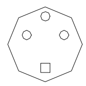
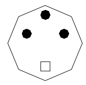
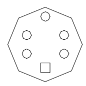
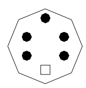
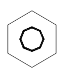
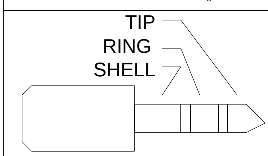
3-Conductor Standard 1/4" I.D. Types

Typical mating type: 12B † 502-60: .28" cable O.D.

MOUSER STOCK NO.	Fig.	Handle	Terminal	1	10	25
502-260	A	Black	Screw	5.32	3.99	3.66
502-267	A	Black	Solder	4.52	3.39	3.11
502-269	A	Red	Solder	4.52	3.39	3.11
502-297	B	Shielded	Solder	5.36	4.02	3.69
† 502-60	A	Black	Solder	4.88	3.66	3.36

On a recent High School auditorium renovation project, the Electrical Contractor asked for clarification of the XLR part numbers specified. The following graphic was added to the construction drawing set.

CONNECTOR SCHEDULE

SYMBOL	DESCRIPTION	SIMILAR TO
	XLR-3 CABLE FEMALE	SWITCHCRAFT A3F
	XLR-3 CHASSIS FEMALE	SWITCHCRAFT D3F
	XLR-3 CABLE MALE	SWITCHCRAFT A3M
	XLR-5 CHASSIS FEMALE	SWITCHCRAFT D5F
	XLR-5 CABLE MALE	SWITCHCRAFT A5M
	1/4-IN TRS CHASSIS FEMALE	SWITCHCRAFT 12B
	1/4-IN TRS CABLE MALE	SWITCHCRAFT 238

Mini-plugs (3.5mm) are not traditional professional audio connectors. However, panel space is becoming a limiting factor on miniature electronics, as wireless microphone transmitter packs. Therefore, the tiny connectors are being used. They are similar in every way to TRS connectors except that the size almost precludes manual soldering.

Microphone Cable:: .

This is the time we must discuss the non-issue of microphone impedance. In theory (and in practice 50 years ago) there are two very different connections available between audio sources and audio loads - high impedance and low impedance. High impedance was an almost direct connection to the grid of a vacuum tube. Low impedance was connection to a step-up transformer which was almost directly connected to the grid of a vacuum tube. High impedance had better frequency response, but was limited to short distances before noise pickup became noticeable.

Today, we use field effect transistors (FETs) for input stages on audio equipment. They provide a high impedance input (low current flow), but work equally well on high impedance and low impedance sources. Theory requires that low impedance sources be terminated - with a transformer or resistor. In rare cases, this is performed. Usually, however, the FET input simply bridges the input and works equally well on high impedance or low impedance sources.

[This discussion applies only to millivolt signal circuits. Impedance matching is still critical on power circuits and high frequency circuits, as video.]

Another non-issue is balanced vs unbalanced lines. Every circuit has a source conductor and a return conductor. In a balanced circuit, they are treated as equals. In an unbalanced circuit, the return is considered second rate and must share its current path with the non-current carrying shield. Unbalanced circuits came with high impedance vacuum tube grid input circuits. Balanced inputs are now provided by push-pull inputs of FET's. If an unbalanced source is connected to a balanced input, the input tries very hard to provide a constant zero signal through the grounded side. Works well. Many mixers, for example, permit selection of balanced or unbalanced input from the same connector.

The question outstanding is, what microphone cable to specify for a sound reinforcement system? Typically, the answer is two - a super flexible, rough-service shielded cable at the mike, and an economic, well shielded cable in the wall to the mixer.

Old and abused mike cables "crackle" when bent, or when the performer walks around the stage. They must be replaced. The author recommends an ethylene propylene rubber (EPR) jacket over braided tinned copper shield with two #16 conductors. EPR hold up well, but can be cut or crushed. Braided shield is a little stiffer than spiral, but has better shielding and heals at broken strands. #16 is much larger than required for the signal, but provides the only strength in the cable. There are many opinions on mike cable and all work well initially.

In-wall mike cable can benefit from modern technology. Broadcast studios use #22 solid or stranded conductors in an aluminized Mylar shield with a tinned copper drain wire. The overall jacket is some plastic. (The jacket must be Teflon if the cable will be open-strung in air handling spaces.) This works because the cable is installed and left forever. Cost is 10 cents per foot. On the school auditorium job used as an example earlier, the author was concerned that the wire puller would overstress the cable. For this reason, #16 conductors were specified. Cost is 30 cents per foot. The cable schedule from that job is reproduced below.

CABLE SCHEDULE (300V JACKET)		
CONNECTOR	CABLE	SIMILAR TO
MIKE	2/C#16 SH	WEST PENN 294
DATA	4-PR CAT 6	WEST PENN 54568
F-FIDEO	RG-11	WEST PENN 1100
AUDIO	2/C#12	WEST PENN 227
DMX	5/C#18 SH	WEST PENN 3280
BNC-VIDEO	RG-59	WEST PENN 840

Preamplifier:
The preamplifier device has become invisible in modern sound reinforcement systems. Look for the term in the description of the mixer. A microphone has an output signal of millivolts, about –60dB. A mixer is optimized for 1 volt, 0dB or 300mV, -6dB. A preamplifier is needed between the microphone and the mixer. Because of the high amplification of a very weak signal, early preamps contributed most of the noise heard on the speakers and deserved close attention. Modern FET preamps have very, very low noise and get little notice. If you must buy a preamp, the cost is $60-150.

As part of incorporating the preamp into the mixer, it is often joined with digital to analog conversion and the first stages of digital processing. This appears complicated. It is complicated. But, in today's market place, it actually reduces the price of the components and improves features and delivered sound quality.

The importance of the preamplifier in system specification is the microphone input count. Each microphone requires one pre-amp. Depending upon how stereo is handled, the mike input count goes to 24 channels very rapidly. For whatever reason, mixer manufacturers like many high level inputs and few microphone inputs. One solution is a matrix switcher, so that many mikes can use few inputs, but this becomes complicated for the user and inappropriate for schools and churches. Mixer channels are cheap, but the boards become huge.

Mixer:
Many years ago, a mixer channel was an audio potentiometer and an isolation resistor. Six channels meant six pots and six resistors. This primitive mixer sells today for $60.

A powered mixer is a primitive mixer combined with a power amplifier. It may also contain some DSP functions. A powered mixer costs $100 to $1000.

A modern mixer must accept microphone and high-level inputs. It should have a master gain. It must have enough amplification to provide zero insertion loss to the system. It is desirable, almost essential to have multiple outputs, for monitor speakers, headphones, recording, assistive listening and digital processing for multiple speaker channels. It should have LED level indicators, preferably on each input. This describes the high-end mixer, introduced below.

High-end Mixer:
Live music performers usually bring along their own sound system and technicians. This clarifies responsibility for proper operation and provides uniformity and confidence to the performers. Their choice is to have many, many microphones on stage, some wireless, many, many wires (snake) to a massive high-end mixer in the first few rows of audience space, dead center, and multiple audio returns back to the stage. The return channels are for performer monitor speakers, to the power amplifiers, to multi-channel recording gear and to the assistive listening system (sic).

The high-end mixer has an input channel for each microphone and source, always with individual level control and usually with individual equalization and digital signal processing (effects) controls and LED level meter. The processed signal from each input channel can be directed to one or more master channels. The master channels have master level controls. A moderate high-end mixer has 24 mike inputs, four master channels and costs $400-$5,000.

Inputs are XLR, TRS, RCA and proprietary connectors. DC power is available for condenser mikes. Outputs are XLR, TRS, RCA and proprietary connectors. The simplest proprietary connectors are high-density DIN separable terminals. The stripped conductors of a mike cable are inserted into the high-density connector, the screws are tightened and the connector is inserted in the mating strip. The remote control unit using these has 12 channels in a 1-1/2-in rack space.

It is hard to get a high-end mixer that will fit in a 19-in audio rack. Many are six feet long.

(**Remote control):**
Current commercial audio technology does not support remote control, that is, digital commands over a single circuit to operate the mixer functions. The function can be approached, however, using a matrix-switch/digital mixer from Mackie Industrial ($1000) or the system processor matrix switch/digital signal processor from Shure ($3000). With either unit, inputs can be selected and master volume controlled by an expendable remote box. This is extremely desirable for shared public venues. The cost is not high for a commercial system expected to perform reliably for 20 years.

Signal Processor:
The signal processor is a digital signal-processing (DSP) chip controlled by a microprocessor and a very sophisticated program. Anything that has ever been done by electronics can be done and extended by DSP technology. The programmers have gone wild. Three functions will be described here - feedback elimination, time alignment and equalization. Feedback, as discussed in the acoustics section, is the natural resonant frequency of the space magnifying any source content at that frequency. If the sound reinforcement gain is high, the single frequency, magnified by the resonance, will cause squeal. A common DSP function is to have the unit find that frequency and reduce the gain for that frequency only. Multiple frequencies can be identified and muted, all automatically.

Time alignment is the constructive response to interfering echoes among multiple speakers. The performer and speakers at the front release a sound. It takes 100 milliseconds for that sound to travel 100 ft into the audience. If an adjacent speaker reproduces the sound when made, then the audience receives two versions, real-time and 100mS delayed. The human ear can detect 30 mS and loses coherence of the message with a loud 100mS interference. If separate amplifiers drive the down front speaker and the 100 ft away speaker, a DSP delay of 100mS can be inserted in the 100 ft away output so that it arrives with the real-time version from the front. Sounds match. Lips way out of sync.

Equalization is the idea that a good technician can tune the sound system to improve the listening experience. A stand-alone equalizer is a row of +/-10 dB sliders, usually at ½ octave centers, for very narrow filters. Each pass band can be boosted or diminished.

A low-end DSP preamp will do all this and much more, for $60. A high-end DSP requires a week of training before use and costs $3,000.

Assistive Listening:
The author's current interpretation of the Americans with Disabilities Act (ADA) is that places of assembly (50 persons or more) must be equipped with an assistive listening system to aid hearing impaired persons. An attorney or architect will be able to provide more competent advice. The requirement appears to be 4% of audience capacity, that is, 4 receivers per 100 seats or authorized standing spaces.

As indicated on the sound system graphic, an assistive listening system is made up of a transmitter, which accepts a line level input, multiple receivers, and chargers for the receivers. Multiple channels are available. The graphic indicates a radio system. Infra red systems are also available, with some serious problems.

An assistive listening kit costs about $800.

Power Amplifier:

There are two distinct products in the commercial sound reinforcement field which are called, “Power Amplifiers”. The first is better termed a “powered mixer”. It contains rotary level controls, rudimentary equalization (tone controls), sometimes a feedback filter, and a power amplifier in the 20-1000w range. Well-known brand names are Bogen and DuKane. Cost is $100-800.

The second category is a raw power amplifier. It usually contains a master volume control and speaker protection. Some contain a slot for one or more microprocessor/DSP module. Brand names are Crown, Rane and Electro-Voice. Cost is in the $400 –4000 range.

A power amplifier for a small installation is a non-issue. As shown by the field data, 1 w rms provides very acceptable sound levels in a space 30ft x 30ft. A 100 w amplifier far exceeds the need of any small space and is an economic price point in all manufacturer’s price list. Because of the excellent noise characteristics of solid-state devices, there is no downside to over sizing the amplifier.

At this point, we begin a concealed discussion of two arcane subjects, speaker damage and speaker impedance. A 200w amplifier is connected to four 200w speakers. If the amplifier rating is conservative, there is still plenty of headroom in the speaker ratings to permit quality, safe operation. Right? Maybe.

The 200w rating on the amplifier is real. Manufacturers use a number of standards, but they all include a rated input signal and rated rms power output into rated load for an extended period with no damage. (RMS power is what you read with a good meter. A cheap meter may read peak or average power.) To get continuous 200w power, the amplifier is inherently capable of producing 400w, possibly 800w briefly. Music tends to contain brief, high intensity passages. Politicians and rock groups tend to favor driving a system to its maximum capabilities (turn it up all the way). At maximum gain, the amplifier distorts and produces high magnitude, high frequency signals which exceed the speaker high frequency capability. One manufacturer stated that his 100w full-range speaker is only capable of 20w at the tweeter. It is sized for the frequency distribution of music and voice.

For this reason, speaker protection is a valuable feature in the amplifier. Also, speakers should be sized conservatively. Another alternative is to put in an oversized amplifier so that it does not distort on peaks. It will be capable of really massive overload signals if turned all the way up.

Speaker impedance contains many nasty details that are better avoided. There are two conclusions - identical speakers will work well together and series-parallel connections are not difficult. Sound from 20 to 15,000Hz contains a lot of frequencies. Speaker manufacturers have 100 years of tweaking up the mechanical and electrical designs to get linear response. It works. A given speaker (tweeter, mid-range and woofer) can be expected to work well. In addition, four of them or eight of them or even 20 of them and they will work well together.

Different speakers, however, have different compromises to produce net linear sound output. Do not expect them to work well on the same output circuit. The will interact badly, though possibly not noticeably to the audience. The solution is multiple power amplifiers if different speaker types must be used together. This is one of the reasons for multiple send channels on a mixer - to drive multiple power amplifiers. If you were studying the discussion of time alignment in the DSP discussion, you noticed that multiple amplifiers are required for that also.

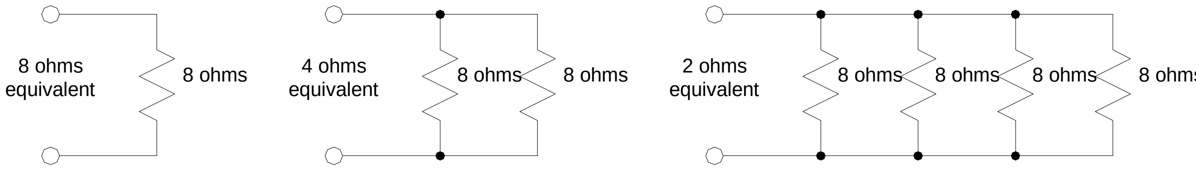
On the other hand, cafeterias, auditoriums and small churches are usually well provided by a reasonable number of identical speakers. No problem.

There are three concepts of connecting speakers to the amplifier that we would like to avoid, but cannot - matching nominal impedance, series-parallel connections to match the direct output and transformer connections for a transformer output. Modern solid-state amplifiers are rated at 8 ohms output, with 4 ohms and 2 ohms in fine print. Most speakers are rated at 8 ohms. The fine print in the speaker specification states that actual impedance ranges from 2-8 ohms, with midrange about 5 ohms. As long as we have plenty of watts in the amplifier, this doesn't matter. Otherwise, get more watts in the amplifier.

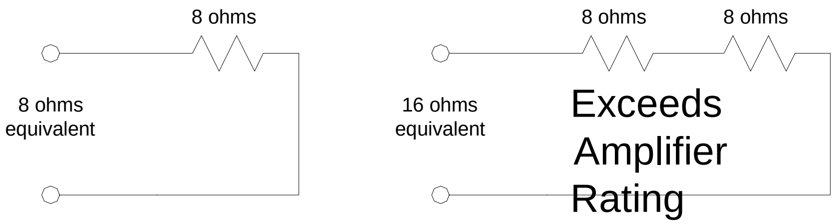
Connect an 8 ohm speaker to the 8 ohm amplifier terminal. Simple, easy no problem.

How to connect two right side speakers to the single 8 ohm out put, though? Put them in series and two eight ohm speakers look like 16 ohms. Put them in parallel and two 8 ohm speakers look like 4 ohms. This is where the fine print on the amplifier specification saves the day. 4 ohms is OK. Four speakers in parallel look like 2 ohms and 2 ohms is OK. In addition, any wiring errors are covered by the output protection in the amplifier.

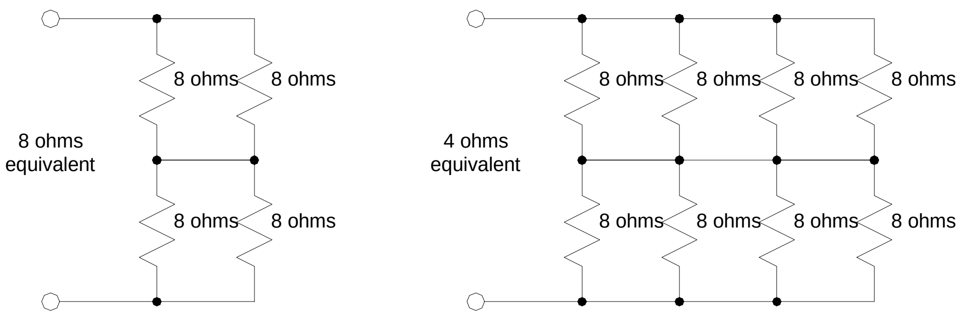
What about more than four speakers on an output? What about trying to connect 8 ohms load to a rated 8 ohm source? That is where series-parallel connections come in. (Series-parallel connections are discussed in the equipment manuals and at the manufacturers' websites, also.) The graphic below illustrates common series parallel connections.



Parallel Connections



Series Connections



Series-Parallel Connections

Good design requires separate conductors from the amplifier to each speaker, with series-parallel connections behind the amplifier. Simplified installation practice suggests daisy-chaining speakers, with connections in the field.

Transformer output is a good idea for the County Fair, where speaker runs are in the 1000's of feet. It is an economical use of equipment and cable and fairly simple to design and install. Unfortunately, frequency response is not music quality and it is just marginally legal.

Direct output matches speaker impedance. It works well because transistor characteristic impedance also matches speaker impedance. Hence, direct output. Unfortunately, 100w at 8 ohms is 28 volts at 3.5 Amps. Three point five amps cause a voltage drop (signal loss) over 1000 ft of wire, even over fairly large, #12 wire. Twenty-eight volts cannot afford much voltage drop.

The solution of 50 years ago is to boost the voltage with a transformer. The line current is proportionately reduced. Voltage drop is reduced and the base voltage doesn't care as much. The downside is the losses and frequency limitations of the transformer and the requirement for a transformer at the speaker, with losses and frequency limitations.

Historically, sound installations have been performed after the Electrical Inspector left the building. Voltages were low and any hazards were considered minimal. This is changing. The 2002 National Electric Code includes low-voltage wiring and requires the same rules apply as with 120V wiring. (This is because of the proliferation of data wiring, but still applies to sound wiring.) Inspectors today are not checking sound installations, but they have the authority and it is an individual decision. Eight ohm wiring meets the low voltage rules. Transformer operated speakers wiring usually does not.

Speaker Wiring:
Two heavy stranded copper conductors with a 300-volt jacket are needed for each speaker. Specify a Teflon jacket if open wiring is run through an air handling ceiling space. The 300-volt jacket provides physical protection and stops the Electrical Inspector from asking questions. Stranded is much easier to work than solid conductors and survives abuse better. Heavy means #18, #16, #14 or #12 AWG wire size. #12 is best and not much more expensive. Cost is 10-40 cents per foot.

Beyond these minimal requirements, many people have many opinions. The preferences are not supported by double blind A-B tests, but it is a good idea to help the Client spend his money as he wishes and not argue.

Eight-ohm speaker wiring can be run anywhere you can drill a hole or drive a staple. Ideally, the wiring is concealed during construction and conduit or surface raceway are used. Sound installers like to drill holes or poke through gypsum walls with a screwdriver and hang the wire in space. Most of the workmanship is at ceiling level or behind the speaker.

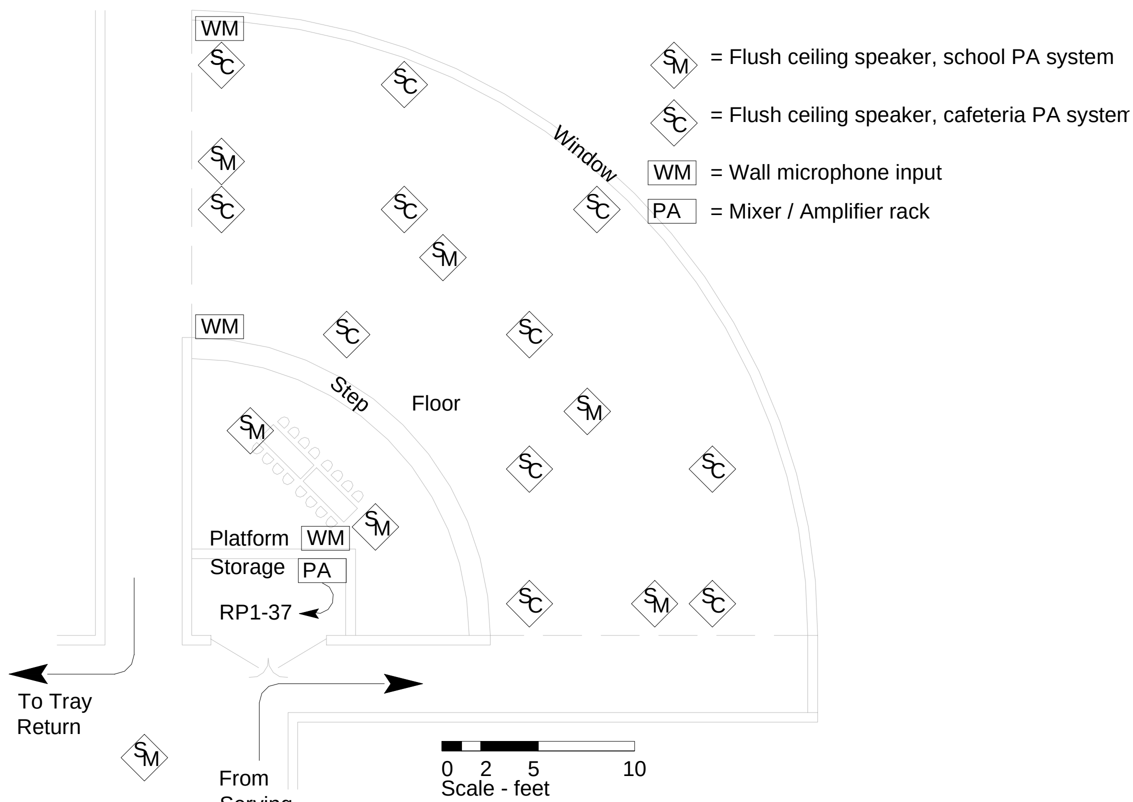
Speaker:
Regardless of the project, cafeteria, auditorium or small church, the speaker requirements are the same. A speaker in front of and on each side of the presenter are needed. Preference is to hang them near the ceiling. After that, several speakers along the sides or distributed over the ceiling.

Details of speaker selection are discussed in the Power Amplifier section. Good speakers cost $100-600 each and should last 10-20 years.

	1950's Sound Design Textbook Recommendations for Many Small Speakers in Ceiling									
	(Standard 30-ft x 30-ft classroom or conference room = 900 sq-ft.)									
Example *	Ceiling Height *	Ceiling Height - 4 ft	Area per 60-deg spkr, sq-ft	Speakers per 1000 sq-ft	Mounting centers, c-c, ft *	Required Watts per 1000 sq-ft*	Area per 90-deg spkr	Speakers per 1000 sq-ft	Mounting centers, c-c, ft *	Required Watts per 1000 sq-ft*
	7.0	3.0	9.0	166.0	2.5		28.0	52.0	4.4	
	8.0	4.0	16.5	91.0	3.3		50.0	30.0	5.8	
	9.0	5.0	28.5	52.0	4.4		78.0	20.0	7.1	
classroom, office	10.0	6.0	38.5	39.0	5.1	40.0	114.0	13.0	8.8	13.0
	11.0	7.0	50.0	30.0	5.8		153.0	10.0	10.0	
	12.0	8.0	65.5	23.0	6.6		200.0	8.0	11.2	
	13.0	9.0	85.0	18.0	7.5		256.0	6.0	12.9	
	14.0	10.0	106.0	14.0	8.5		314.0	5.0	14.1	
cafeteria	15.0	11.0	128.0	11.0	9.5	50.0	380.0	4.0	15.8	25.0
	16.0	12.0	154.0	10.0	10.0		452.0	4.0	15.8	
	17.0	13.0	176.0	9.0	10.5		530.0	3.0	18.3	
	18.0	14.0	205.0	7.0	12.0		620.0	3.0	18.3	
	19.0	15.0	235.0	6.0	12.9		710.0	2.0	22.4	
	20.0	16.0	271.0	6.0	12.9		810.0	2.0	22.4	
	21.0	17.0	308.0	5.0	14.1		910.0	2.0	22.4	
party center	22.0	18.0	345.0	4.0	15.8	100.0	1020.0	2.0	22.4	50.0
					* = column added to original table					

SAMPLE CAFETERIA SOUND DESIGN:
A sound system plan drawing is usually sparse. That is, the locations of devices are indicated roughly, for field fit into the ceiling grid, etc. The devices themselves are indicated very generally, without detailed layout or wiring. The components may be precisely described on accompanying specifications, but the specifications are ignored by the supplier and Construction Manager. This is because each supplier has a close relationship with one or a few manufacturers and gets substantial discounts on only these items. Also, the sound installer has substantial incentive to install reliable components - the meetings are more expensive than the components and the installation. A meeting with a principal of the sound firm will cost $100 per hour. The focus of such meetings must be on up selling the customer and positioning the firm for future jobs. Discussion of 5-watt or 10-watt ceiling speakers distracts from the sound person's purpose. He will personally guarantee proper operation and go to lengths to deliver. Note that any sophisticated input control will be provided by portable equipment with a microphone level output.

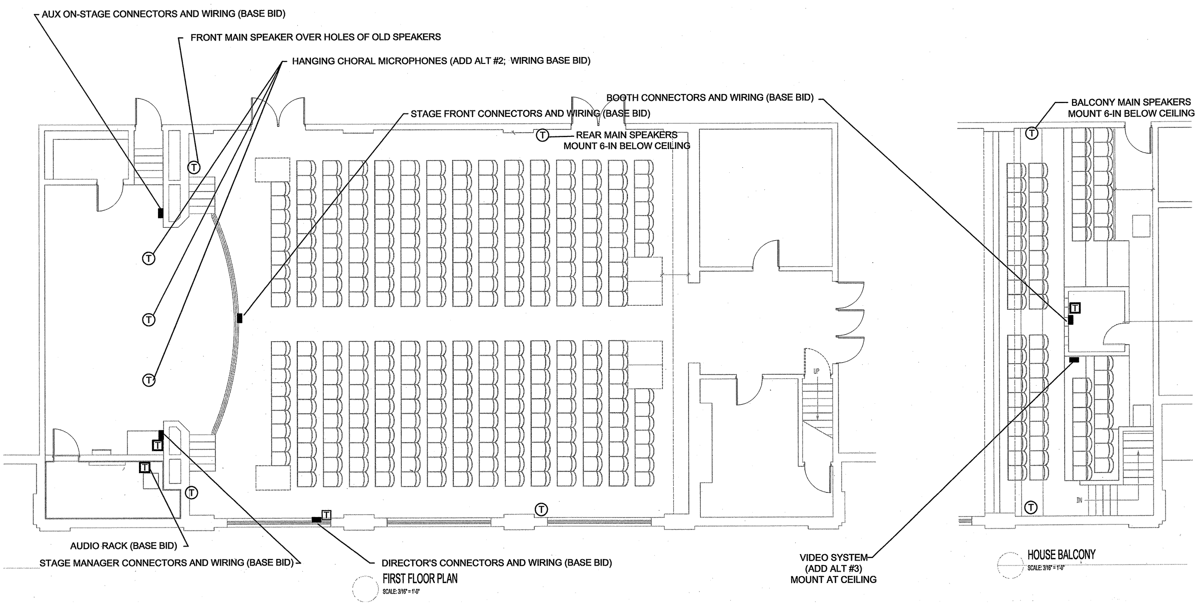
Some installers are following the National Electrical Code and using drive rings and bridle rings for cable support.



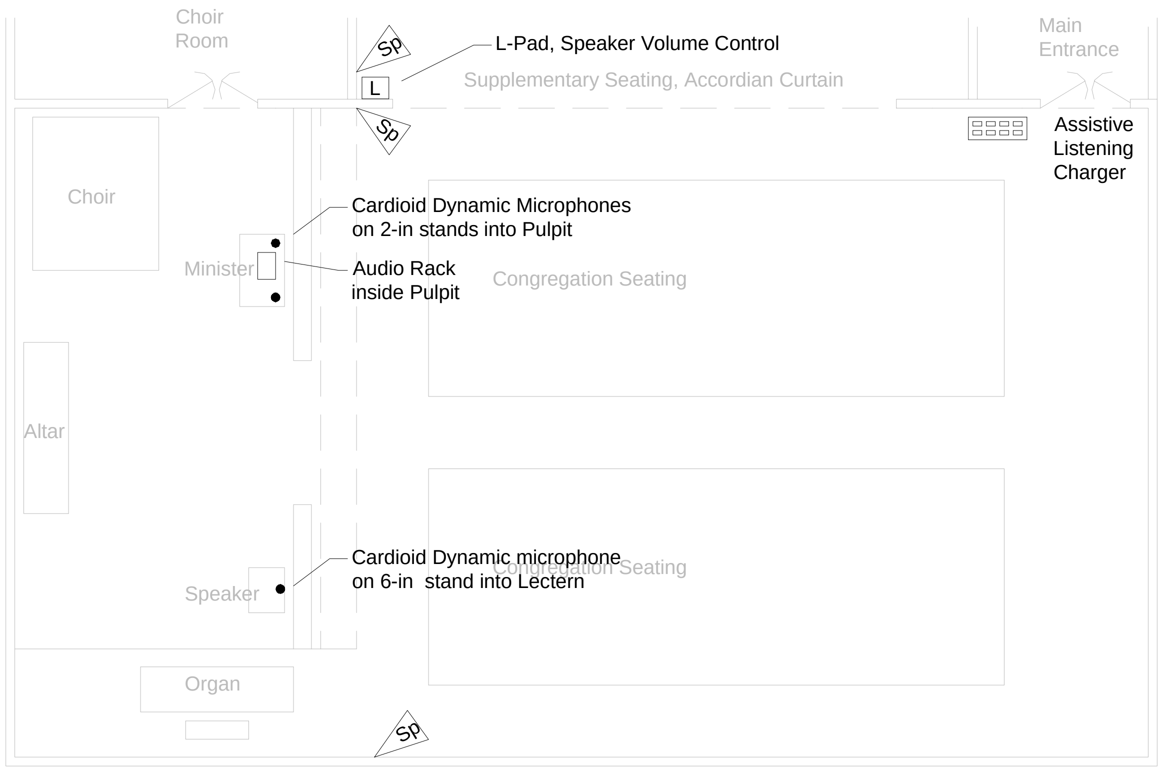
Cafeteria Sound System Plan

SAMPLE AUDITORIUM SOUND DESIGN:
Like the cafeteria sound design, the auditorium plan drawing is sparse. The dimensions of a high school auditorium approach the echo interference limit, where sound alignment processing is required. The base design for this auditorium used two primitive mixers for additional channels and a 100w powered mixer, driving six 400w speakers in series-parallel. An alternate deleted the powered mixer and added a dsp/matrix switch and two 400w power

amps. The reason for the oversized speakers in the base was so that the upgrade could be made later if the alternate was not accepted.



SAMPLE SMALL CHURCH DESIGN:
Like the previous sound designs, the small church plan is sparse. The amplifier was a 100-w powered mixer. The speakers were music-rated 100-w corner enclosures. The rack contained the transmitter for the assistive listening, including the antenna. The speaker volume control has not been discussed previously because it is a high-maintenance item, to be avoided in school applications. It is important to get an L-pad rated for the expected power level. A 5w L-pad worked well here.



Small Church Sound System Plan

COMMON DESIGN ERRORS:

1) It is critical that the relative locations be as follows:

Performer or musician Speaker(s) Audience.

If the speakers are behind the performer, available gain is severely limited, so that soft-voice
persons will not be heard. Increased gain will produce feedback.

Speakers behind the audience produce an eerie discomfort in the audience.

2) More small speakers are preferable to few large speakers.
A ceiling-full of flush mid-range speakers on 15-ft centers will produce uniform sound. Four or six speakers along a wall can be driven at low level. Listeners close by hear well; persons further away do not get interference.

3) Talk to the installer. Typically, the school or Owner has a sound supplier who has done most of the work for them over the years. Use his accumulated knowledge on how to make the Owner happy. Talk to him about the scope of the project and features needed. Give him a chance to review and mark-up the design. Most sound designs are created by the supplier and copied into the project set by the Engineer or Architect. Asking for review and comments is much more ethical. Keep “or equal” in the manufacturers list.

4) Serious acoustics are complicated, not intuitive. A medium-sized Ohio city has a performing arts hall. For twenty years, the built-in sound system produced pure garble in the balcony. Since those are the cheap seats, no action was taken. It is amazing to some that a rented venue can survive with bad sound.

   Attention to acoustics is obvious. There will be adjustable reflective and absorptive surfaces (doors). There will be nothing prominent about the sound system except that the audience experiences vivid clarity of speech and music. A trained, experienced acoustician costs $1000-2000 per day for opinions, not a design. Acceptance testing of the sound panel adjustments requires a full audience for the tests and recordings rather than expert opinion.

5) There is value to simplicity. A school sound system must be operable by the Principal. That means one switch and plugging in the microphone. A high-end mixer is wonderful for a trained operator, but one of the channels, or a bypass, must be dedicated to ad-hoc assemblies.

COMMON INSTALLATION ERRORS:

1) Installers want to install the brand they get the largest discount on. This provides competition and benefits to the project budget and to the Owner. It is essential for the designer to check the specifications of the materials offered. Specify the microphone pickup pattern; check the microphone pickup pattern. Specify the microphone channels; count the microphone channels. Specify the rms power rating; check the rms power rating. Specify the speaker wire size; check the speaker wire size. Do not accept, “This will work just as well.” Identical specs are required to work just as well.

2) Talk to the installer about workmanship before he starts on the job. Emphasize the need for aesthetics, especially in the cable exits from the wall or ceiling. The right way is a junction box or back plate fitting, and a cover plate with a gland fitting for cable strain relief. Installers like to poke through a big hole with a screwdriver and let the cable hang. This is a violation of the National Electric Code and really ugly when you focus on it.

3) Terminations are the part of all electrical installations most prone to error. Usually, a bad audio termination is obvious when you turn up the volume and listen for hum. Wiggle the wires at the audio rack. Deafening clicks and even a little hum are indications of poor terminations. The deafening clicks are destructive to speakers, so have an installer technician do the wiggling.

4) Remember that an acceptance test requires a full audience. The space acoustics change considerably.

PLAN AND RISER REPRESENTATION:

A sound reinforcement system plan diagram is a building plan with locations of sound system devices shown by generally recognized symbols. At present, there is no consensus standard for the symbols. Use symbols of your choice, but include a legend. The plan drawing should always be accompanied by a Symbol Legend.

The building plan, called a "background," is provided by the Architect, who also determines the use group, construction, surface treatments and provides final acceptance of speaker locations. The designer takes on substantial liability if he makes assumptions written direction or sign-off of preliminaries from the Architect.

Three sound system plans were presented previously.

A riser diagram is a graphic Bill of Materials for the job. It shows all essential components, essential specifications and calls out count. It should match the specifications exactly. Often no one reads the specifications. The introductory sound system graphic for this course illustrates a sound system riser diagram, but lacks the ratings call-outs.