# Sound Reinforcement for Speech

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# Contents

# Chapter 1

# Introduction

This document has been prepared as a resource to help you set up, improve, maintain, and operate sound systems for speech reinforcement.

The primary focus is on the theory, so you can apply it in your particular situation with the hardware you already have. Hardware suggestions are also offered. If you would like a ready-to-use complete system design instead, without the theory, please refer to the “Speech Reinforcement Design” documents available on owenh.net.

This is a **work in progress**. This document is provided with no warranty or guarantee of any kind. If you find any error here and can support an argument against it, please contact me. “Thank-you” to those who have answered my questions or otherwise offered input.

## 1.1 Signal path

A sound system is a collection of interconnected equipment, and the audio signal travels through each of those parts. The performance of the system as a whole is affected by each part of this signal path. Each chapter in this book discusses a particular part of the signal path. **To keep your signal quality high, every component in the signal path needs to be working correctly.**

Here is an example of a signal path:

**Human speaker** → microphone → *mixing/processing* →

1. → Room speakers → room → **ears**
2. → assistive/outside listening → **ears**
3. → telephone network → **ears**

## 1.2 Selecting hardware

When selecting hardware, look for quality first, then look for a feature set close to what you need. Often a well-designed device with a limited feature set will work better than a poorly-designed device with an extensive feature set. Do not make the mistake of looking for the cheapest product with the features you need: likely those features are poorly implemented.

If price is a concern, consider buying used equipment. Generally, used high-quality hardware will serve you better than new low-quality hardware. But remember to test used hardware thoroughly before using in an important event!

## 1.3 Other considerations

- See the Wiring and Cables chapter for information about cable types, distances, and gauges.
- See the Power chapter for information about powering your sound equipment.
- See the Storage storage chapter for information about storing your sound equipment.
- See the Hardware Brands chapter for some brand recommendations.

## 1.4 Todo

- http://www.rane.com/note163.html
- http://www.symetrix.co/products/airtools/
- http://www.rane.com/note162.html
- https://shure.custhelp.com/app/answers/detail/a_id/2972/~/automatic-gain-control-%28agc%29-in-shure-dsp-products
- http://www.rane.com/note155.html
- http://sound.westhost.com/articles/line-amps.html
- http://primacoustic.com/app-house-of-worship.htm
- http://www.prosoundweb.com/article/print/in-depth_primer_on_speech_intelligibility_analysis
- http://www.meyersound.com/support/papers/speech/section2.htm

# Chapter 2

# Human Speaker

The human behind the microphone is obviously the most important part of the signal path. A well designed sound system can easily accomodate most voices.

## 2.1 Suggestions for the person speaking

It is seldom practical to try altering the behavior of the person speaking, and generally impossible to affect a change in the actual sound produced. However, there are a few suggestions that could be shared with any speaker who is interested.

1. It can be counter-productive for the lips to be closer than 3 inches from the microphone. This is always a compromise between gain-before-feedback and signal accuracy. ONLY when feedback is a problem should the speaker move closer to the microphone (and feedback should not be a problem with proper speaker placement and managed frequency response). For more information about the proximity effect, see the Microphones chapter.

2. Speak clearly to establish a high initial SNR. TODO speaking loudly, Lombard effect

## 2.2 Frequency ranges

Having covered some suggestions for the speaker, we now are concerned with the technical characteristics of voice so we can understand what our "signal" actually is.

- Typical adult male speech fundamental frequency: 85 Hz to 180 Hz.
- Typical adult female speech fundamental frequency: 165 Hz to 255 Hz.
- Child's speech fundamental frequency: 250 Hz to 300 Hz or higher.
- Maximum vocal range: 65 Hz (male) to 1.28 kHz (female).
- Sibilance, or essing, is the high frequency sound associated with the letter s. Sibilants range from 2 kHz to 10 kHz.
- Speech critical range: approximately 170 Hz to 8.3 kHz.
- Range for high-fidelity speech reproduction: 80 Hz to 12 kHz.
- In telephony, the usable voice frequency range is approximately 300 Hz to 3.4 kHz. This is not adequate for high-fidelity speech reproduction; the fundamental frequency of most speech and much of the sibilance of speech falls outside of this range.
- As comparison, a standard piano keyboard covers tones from 27.5 Hz to 4.186 kHz.

Sources:

- Wikipedia: Voice frequency
- Axiom Audio: Frequency Ranges of Male, Female, and Children's Voices

### 2.2.1 Speech critical bands

The following is a selection of narrow frequency bands which may be treated as the elementary signals in speech.

Table 2.1: Speech critical bands

| Center | Width | Range (Hz) |
|---|---|---|
| 200 | 60 | 170-230 |
| 300 | 60 | 270-330 |
| 500 | 60 | 470-530 |
| 800 | 70 | 765-835 |
| 1000 | 80 | 960-1040 |
| 1500 | 100 | 1450-1550 |
| 2000 | 130 | 1935-2065 |
| 3000 | 200 | 2900-3100 |
| 5000 | 300 | 4850-5150 |
| 8000 | 600 | 7700-8300 |

Consequently, on a 1/3 octave equalizer, the following bands are especially important (though everything from 80 Hz to 10 kHz is important for high-fidelity reproduction):

- 160 Hz
- 200 Hz
- 250 Hz
- 315 Hz
- 500 Hz
- 800 Hz
- 1 kHz
- 1.6 kHz
- 2 kHz
- 3.15 kHz
- 5 kHz
- 8 kHz

Sources:

- denoise.net: speech bands

## 2.3 Notes/Todo:

- De-essing
- https://en.wikipedia.org/wiki/Intelligibility_%28communication%29

# Chapter 3

# Microphone

The microphone is the first component in the signal path that you have full control of. You need a good one, one designed specifically for speech. Other traits include: type, pickup pattern, frequency response, and proximity effect.

## 3.1 Consider These Microphones

### 3.1.1 Dynamic

- Sennheiser e835: "Dynamic cardioid microphone designed for speech and vocals."
- Shure SM48-LC: Dynamic cardioid microphone "for lead vocals, backup vocals, and spoken word applications".
- beyerdynamic M59: Hypercardioid dynamic microphone for "in-studio or on-location ENG/EFP speech and interview applications."

TODO: condenser phantom-powered, condenser battery-powered

## 3.2 Microphone types

- **Condenser** microphones are sensitive and provide a high-quality signal. This is often especially noticeable in the higher octaves where sibilants occur, thus delivering improved speech clarity. Condenser microphones require phantom power, usually provided by the mixer but in some designs by a battery in the microphone. They are generally considered to be more fragile to physical abuse than dynamic microphones.
- **Electret condenser** microphones are permanently charged condenser microphones, but often require phantom power for an internal pre-amplifier. Electret microphones are cheap to mass-produce, and consequently are often cheaper but also of lesser quality than non-electret condenser microphones.
- **Dynamic** microphones are more durable but less sensitive than condenser microphones. They do not require phantom power, but will usually tolerate it without any problems.

Sources:

- Wikipedia: Microphone
- Wikipedia: Ribbon microphone

## 3.3 Polar (pickup) patterns

- **Omnidirectional** microphones pick up equally from all directions (bit a bit less from the rear). They do not suffer from proximity effect.

- **Subcardioid** or **wide cardioid** microphones are only slightly more directional than omnidirectional and slightly less directional than cardioids. They exhibit very little proximity effect.
- **Cardioid** microphones have a heart-shaped (cardioid) pickup pattern, with more directivity than the subcardioid. Cardioid microphones suffer from proximity effect.
- **Proximity effect** is a result of the design of directional microphones. It causes the frequency response of the microphone to change, boosting the low range significantly as the sound source gets nearer to the microphone. See Shure answer 2844: "Why does Proximity Effect Occur?". For a plotted example of the proximity effect, see the Shure Beta 57A specifications.
- **Supercardioid** and **hypercardioid** pickup patterns are too focused for most speech applications. It is far worse to discard valid signal because the speaker is a bit off-axis than to pick up some noise (which is at a much lower level anyway). Generally, do not attempt to solve feedback problems by using more focused pickup patterns.

Sources:

- Wikipedia: Microphone Polar Patterns
- Shure: Multi-Pattern Microphones: What, Where and How?
- Sound On Sound: Recording The Spoken Voice

## 3.4 Microphone stands

### 3.4.1 Basic

- On-Stage MS7700B

### 3.4.2 Portable

- goSTAND Portable Mic Stand
- Manfrotto 5001B
- Manfrotto 1051BAC
- Lowel Uni-TO
- On-Stage MSA-9500
- Shure S15A
- Avenger a635b
- Avenger a0035b
- Auray MS-5220T
- camera tripod stands?

### 3.4.3 Remote motorized

- Optogate Mic Lift V2: If I am reading the specifications correctly, this stand provides 72 cm (28.3 inches) of motorized adjustment!
- Chapman Remote-Mast stand model KH, KHW, or KHWB: Unfortunately this unit provides only 30 cm (12 inches) of motorized adjustment.

## 3.5 Todo

- Heil Sound PR-20: of interest. Dynamic cardioid microphone with a moderately flat frequency response, a very reasonable price, and good reviews. I am not sure how much the proximity effect affects the frequency response of this microphone.
- Wired vs wireless
- Shure answer 102: Predicting speech to background noise level at the microphone

- voice processors: dbx 1046, Drawmer DL441, Drawmer DL241, Drawmer MX50-Pro, Drawmer DL251, Drawmer MX30-Pro, Rane DC22S, Samson S-com 4, Samson S-com plus, Symetrix voice 2x and 528E
- USB interfaces: M-Audio M-Track Eight, Tascam US-16x08, PreSonus Studio 192, PreSonus AudioBox 1818VSL
- Shure MX412/C & MX418/C
- cables: Monoprice product #601403, “3ft Cloth Series 1/4 inch TS Male 20AWG Instrument Cable - Black & Gold”
- On-Stage Stands MY200 universal microphone clip

# Chapter 4

# Mixing/processing

## 4.1 Feedback suppressors

- PolyFusion 755: recommended. The 755 may be difficult to obtain, but it is the only device I know of that uses simple frequency shifting instead of narrow notch filters (and is marketed as a suppressor). See the Rane publication "Understanding Acoustic Feedback & Suppressors".

## 4.2 Equalizers

- Peavey CEQ280a: of interest. This is a 29-band equalizer with additional automatic room equalization and pink noise generator. It is simple and seems to work well. Always disconnect the RTA input before live use: on multiple units a connection (balanced cable only, no microphone) there has caused noise on the equalized signal.
- Lectrosonics EQ1: of interest. The EQ1 might be a possible (better?) alternative to the CEQ280a, but it is not easy to obtain.
- Alesis DEQ830: of interest, but *not as an analyzer*. The DEQ830 is great as an equalizer (8 channels!) but the models that include an analyzer provide no range adjustment.

## 4.3 Todo:

- 3 design options: (1) classic analog (may be digital internally),

(2) digital signal processor (DSP) configured by software, (3) processing by computer with audio interface

- noise gates
- compression
- mixers: Alesis MultiMix 12R, Ashly MX-508, Inter-M LM-6414, Rane MLM103, Rane SM26S, TOA M-633D, TOA M-864D
- automixers: AKG DMM series, audio-technica AT-MX341b, audio-technica AT-MX351a, Clockaudio MR-88, Dan Dugan line, Rane AM2, Shure SCM810, Shure SCM820
- DSPs: Atlas IED Audio BlueBridge, beyerdynamic Coretis, Biamp Vocia, ClearOne CONVERGE systems, IVIE iFlex, Lectrosonics ASPEN series, Lectrosonics DM series, Symetrix Jupiter series, Symetrix Solus series, TOA DP-K1

# Chapter 5

# Room speakers

## 5.1 Selecting speakers

## 5.2 Placing speakers

## 5.3 Todo

- Wire
- Placement/coverage
- Quantity
- Feedback
- 70V outputs for inside building, 2 channels
- 70V amplifiers: TOA DA-250DH, TOA DA-250FH
- 70V transformer: TOA MT-251H
- Speaker wire needs no shielding or twisting (except for powered speakers?)
- Speaker wire size should be adequate. This allows for adequate damping, etc.
- http://www.paging-solutions.com/charts/70vwiring.pdf
- https://en.wikipedia.org/wiki/Speaker_wire#Wire_gauge
- http://www.rane.com/note136.html
- http://www.rane.com/note159.html
- installed speakers: JBL
- JBL Rapid DDS
- TOA SR-H2L line array speaker
- TIC SP70vTA outdoor transformer (2.5, 5, 10, 15, 20W taps)

# Chapter 6

# Assistive/outside output

Consider the following requirements:

- easy-to-use battery-powered receivers with headphones for use in the listening area
- same receivers as above for use outside the listening area
- AC-powered or line-powered (passive) permanent speakers for small rooms outside the listening area

## 6.1 Options

- 70 V audio line installed to all required locations for permanent speakers
- Wireless 72 MHz FM transmitter and receivers
- Wireless 216 MHz FM transmitter and receivers
- Wireless 900 MHz FM transmitter and receivers

Notes:

- 70 V is common, and requires a physical line to each speaker. Power is provided by an amplifier in the main sound system.
- One or more of 72, 216, and 900 MHz for assistive listening, may be able to use a single antenna (75*12=900) by isolating grounds of transmitters (both power and input signal). **More research is necessary. Don't destroy any transmitters.**
- The 72, 216, and 900 MHz wireless options require antennas to be mounted. Use good coax, match the impedance, measure your antenna to the correct length, and watch for standing waves.
- 72 MHz provides 6 channels, 216 MHz provides 3 channels.
- Because 216 MHz is a smaller wavelength, it requires a smaller transmit antenna than 72 MHz.
- All other things being equal, 216 MHz will provide approximately double the range that 72 MHz provides. But 216 MHz will lose more power due to cable attenuation than 72 MHz (again, use good coax).

Sources:

- Listentech: Should I Use 72 MHz or 216 MHz?

## 6.2 Todo

- Generic equalizer curve for rooms
- Williams Sound

# Chapter 7

# Telephone

This chapter covers the use of a phone and a conference calling service provider to share the audio from a sound system to remote listeners.

Note that there are also many other ways to implement audio streaming, especially over the internet (such as with Asterisk and MusicOnHold streaming).

## 7.1 Connecting an output to a phone input

Use an XLR to TRRS adapter to connect a system output to a cell phone's microphone input. kV Connection sells two products for this, the KM-IPHONE-MICX and the KM-IPHONE-MICX-A22. These products are identical except that the second one attenuates the signal by 25 dB.

Most outputs will be line level rather than microphone level, and in that case the **KM-IPHONE-MICX-A22** is the appropriate option. If the signal level is still too high, or to use the **KM-IPHONE-MICX** with a line level output, add the Audio-Technica AT8202 adjustable in-line attenuator (selectable 10, 20, or 30 dB attenuation).

Other options: JK Audio BlueDriver-F3, JK Audio Daptor Three

## 7.2 Using FreeConferenceCall.com

**This service is no longer recommended because it provides free conferencing through the use of traffic pumping. This can result in unexpected blocked calls, dropped calls, and usage charges. See related FCC and T-Mobile pages.**

*Dial strings in this section use a p to indicate a pause, sometimes dial string pauses are indicated by commas (,). You should be able to save the entire dial string, including pauses, to your phone's contacts database.* ***Always listen while your phone is dialing to verify successful connection and mode selection.***

1. Register at FreeConferenceCall.com.
2. Record your **dial-in number**, **access code**, and **host PIN**.
3. To provide program audio to the conference, use this dial string:
   <dial—in number> pp <access code> #pppp <host PIN> #ppp∗5p∗5p∗8
   (The ∗5∗5 switches to "presentation mode", with all guests muted; the ∗8 disables the tones indicating when guests enter or exit the conference.)
4. Guests can be given the **dial-in number** and **access code**; they will be prompted to enter the access code and then press the pound/hash button (#). Here is a dial string:
   <dial—in number> pp <access code> #.

# Chapter 8

# Room

The room greatly affects the audio signal. This can be fixed, in part, by installing acoustic treatment. It can be accomodated, in part, by applying to the signal an inverse of the room's frequency response.

Room acoustics, modes, and responses are perhaps the most complicated part of the signal path. This page is certainly not comprehensive, but aims to touch on some basics.

## 8.1 Acoustics

- In a reverberant room (acoustically "live"), a voice can be easily heard throughout the room, but it is difficult to understand. This is because the sound bounces around in the room, so there are many sound waves present based on the original voice, but the timing is all different and the sounds are all colored by the surfaces off which they have reflected.
- In a non-reverberant room (acoustically "dead"), a voice is more difficult to hear throughout the room, but what is heard is easier to understand. In this case there are much fewer reflections, but most of what is heard has come directly from the source.

Sources:

- Acoustics & Sound Systems in the Contemporary Church

## 8.2 Todo

- Feedback suppression using frequency shifting: shift singing up about 2 Hz (or not at all), shift speech down 6 Hz, or shift everything up 3 Hz
- reverberation/RT60: Studios designed for recording speech typically have a reverberation time of about 0.3 seconds (Source). For the human voice, the most critical frequency range with regard to reverberation is 500 Hz to 2 kHz, and carpet on the walls is good at absorbing energy in this region.
- RTA
- acoustic tiles: type, placement, frequency ranges
- ringout correction
- pink noise correction with target curves
- Understanding Acoustic Feedback & Suppressors
- Increasing delay
- Home Studio Corner: Acoustic Treatment vs. Digital Room Correction
- http://www.minidsp.com/applications/auto-eq-with-rew
- http://www.engineeringtoolbox.com/reverberation-time-d_724.html
- http://www.engineeringtoolbox.com/acoustics-noise-decibels-t_27.html
- EASE Focus 2

- EASE Address
- http://www.hometheatershack.com/forums/rew-forum/11707-room-eq-wizard-rew-information-index-links-guides-technical-articles-plea
html
- http://primacoustic.com/app-house-of-worship.htm
- Target 65 dB SPL, reasonable range 60-70 dB SPL

# Chapter 9

# Ears

## 9.1 Frequency

- Hearing range of people with exceptional hearing: 20 Hz to 20 kHz
- Noise-induced hearing loss usually occurs as a notch primarily at 4 kHz and to a lesser extent 3 kHz and 6 kHz, then later at lower frequencies (500 Hz, 1 kHz, or 2 kHz).
- Age-related hearing loss (presbycusis) occurs from 4 kHz to 8 kHz. If you do not have hearing loss, you can hear simulated mild and moderate hearing loss by listening to the samples at the Henry Ford Hearing Loss Simulator (best with earbuds or earphones).

Sources:

- Wikipedia: Noise-induced hearing loss
- Wikipedia: Sensorineural hearing loss

## 9.2 Delay

- Any sound that reaches our ears within 50-80 milliseconds of the original sound gets interpreted by our brains **as** the original sound. Consequently, sound received during that timeframe damages the quality of the original sound.

Sources:

- Home Studio Corner: Acoustic Treatment vs. Digital Room Correction

## 9.3 Todo

- equal-loudness contour, a-weighting

# Chapter 10

# Wiring and Cables

## 10.1 Mic/line level cable

Use a 2 conductor cable with a shield for any of the following:

- microphone level (eg. 0.002 Vrms)
- -10 dBV consumer line level (0.316 Vrms)
- 0 dBu professional reference level (0.775 Vrms)
- 0 dBV consumer reference level (1 Vrms)
- +4 dBu professional line level (1.228 Vrms)

Use eg. Belden 9145 for installed wiring (in a building or rack cabinet).

## 10.2 8 ohm speaker wire

As speaker wire, use 2 conductor wire. Shielding is not necessary and increases the risk of a short.

## 10.3 70V speaker wire

For 70V, use 2 conductor wire. Shielding is not necessary and increases the risk of a short.

18 AWG wire will yield less than 5% power loss up to about 100 m (330 ft).

For installed 70V audio, use 18 AWG or larger wire. Solid and stranded wire are both acceptable, up to 12 AWG. Stranded may be easier to work with when soldering connectors, especially with large gauge wire.

For portable 70V audio, use stranded 18 AWG wire or larger. Stranded wire is more durable for handling.

## 10.4 Attenuators & isolators

- AudioTechnica AT8202
- Shure A15AS
- Whirlwind ISOXL

## 10.5 Todo

- Balanced analog audio, use red/white=positive, black=negative
- Line levels
- Cable shielding
- Digital audio
- Processing sample rate
- Euroblock/Phoenix, use 5.08 mm pitch
- Shure - Should I match impedances of my microphone to my mixer?
- SoundOnSound - Understanding Impedance
- http://www.belden.com/docs/upload/Speaker-Cable-Selection-Guide.pdf

### 10.5.1 Compatible wiring scheme

Amplifier output is 70V, 2 channels, with male twist connector.

Supported outputs:

- 2x 70V bare wire
- 2x 70V phone male (for female jack)
- 2x 70V female twist speaker connector (for male jack)
- 2x 8-ohm bare wire
- 2x 8-ohm phone male (for female jack)
- 2x 8-ohm female twist speaker connector (for male jack)
- 2x 600-ohm (line/mic level) phone/XLR male (for male jack)

Standard cables/adapters required:

- 3x XLR-F to XLR-M standard microphone/line cable, 50 ft (15 m), 16-24 AWG, stranded: TecNec XLM-XLF-50, Audio-Technica AT8314, Monoprice 4756
- 3x speakON-F to speakON-F standard speaker cable, 50 ft (15 m), 12-18 AWG, stranded: Technical Pro CSS1650, Monoprice 8771
- 2x XLR-F to 1/4 inch male TRS microphone/line cable, short, 16-22 AWG, stranded: Monoprice 4767
- 2x speakON-F to speakON-F standard speaker cable, 6 ft (2 m), 12-18 AWG, stranded: Monoprice 8768
- 1x speakON-M to speakON-M splicing adapter (Neutrik NL4MMX): Markertek NL4MMX

Special cables/adapters required:

- 2x 30W 70V transformer/splitter box with 3 parallel speakON-M chassis inputs and 1 speakON-M chassis output: Atlas Sound T20, OSD Audio, Polycase DC-34PMBYR, 9/64 inch bit, 7/8 inch bit, (add template)
- 2x speakON-F to bare speaker wire cable, short, 12-16 AWG, stranded
- 2x speakON-F to 1/4 inch male TS cable, short, 12-18 AWG, stranded

Suggested connectors:

- speakON-F cable connector: Neutrik NL2FX (Markertek), Monoprice 601500
- speakON-M panel mount: Neutrik NL4MP-ST (Markertek)
- speakON-M cable connector: Neutrik NLT4MX (Markertek)

### 10.5.2 More speaker wiring

There are two options for speaker wiring: low impedance and high impedance. Low impedance is typical for basic systems with few speakers and short cable runs. High impedance or constant voltage wiring (typically 70 V in the United States) requires a transformer at each speaker but offers numerous advantages over low impedance:

- All speaker transformers are wired in parallel.
- Higher gauge (smaller) wire can be used.

- Each speaker can use whatever wattage is necessary for its location. The transformers generally have taps ranging from 0.25 W to 30 W, depending on the transformer, to facilitate this.
- The level of a speaker or group of speakers can be adjusted after installation by using a switch-selected transformer and/or attenuator.
- If one or more speakers fail, if the wattage tap/level for a speaker is changed, or if speakers are added or removed, the other speakers continue to operate as before.

See also:

- Rane: Constant-Voltage Audio Distribution Systems
- Wikipedia: Constant-voltage speaker system

Sources:

- Parts Express: Constant Voltage Audio Systems For Beginners

# Chapter 11

# Power

## 11.1 Uninterruptible power supplies (UPSes)

I recommend the **CyberPower OR700LCDRM1U**.

- The battery will charge even when the unit is off. This allows you to use power button on the front of the unit to turn the entire system on/off without concern about whether the battery will be fully charged.
- The *on-battery* alarm can be silenced by holding the select button for 3 seconds. This does *not* disable the following alarms: *low-battery* (seems to be activated when only 5 minutes of battery time is remaining), *overload*, *UPS-fault*, and *replace-battery*.
- It is a line-interactive topology UPS providing automatic voltage regulation (accepts input voltage 90-140 VAC). Line-interactive UPSes do *not* tolerate overloads (it will power off with a long beep).
- You can read input/output voltages, estimated run-time, load capacity, and battery capacity from the display and via USB/serial.
- 700 VA / 400 W
- Replacement batteries: CyberPower RB0690X2. This is actually a combination of 2 batteries, each 6V/9AH with F2 style terminals, together measuring 8.15 x 1.38 x 8.86 inches. Batteryspec.com offers a 2 year warranty and a reasonable price on their compatible TBC113+ (APC RBC18+ compatible) kit with two TR9-6A batteries.
- Manual (PDF)
- Datasheet (PDF)
- Product page

The following models are similar: OR500LCDRM1U, OR1000LCDRM1U, OR1500LCDRM1U, PR750LCDRM1U, and PR1000LCDRM1U.

# Chapter 12

# Storage

## 12.1 Portable racks

I recommend the **Gator Pro Series Molded Racks**, available from 2U to 12U, eg. the G-PRO-6U-19.

- Front and rear lids provide full protection for travel.
- Front and rear rails double the rack space for shallow hardware or provide additional mounting options for longer devices. In the past the product had front rails only and included a $10 offer for rear rails, now the rear rails are included.
- 19 inch depth accommodates longer devices (eg. the TOA DA-250FH).
- Non-wheeled products in this series have smaller external dimensions (total volume) than equivalent products listed below.
- Series page

Other options include Gator Standard Audio Racks, SKB Roto Racks, and Citronic LLDPE 19” Rack Cases.

# Chapter 13

# Hardware Brands

This chapter lists some common brands with notes about their subjective reputations. You can learn more by testing devices or by reading reviews online.

- TOA: recommended. TOA's products, while often more expensive than competitors', are more reliable and have (sometimes radically) better performance. I have had excellent experience with TOA's warranty/repair service.
- Williams Sound: recommended. Their assistive listening systems provide high quality audio transport. This opinion is based mostly the experience of other people with some of their 72 MHz systems.
- Telex: of interest. Decent 72 MHz assistive listening systems.
- Revolabs: of interest. Revolabs has some nice-looking wireless microphone equipment.
- Peavey: not sure.
- Rane: not sure. Rane has some excellent RaneNotes reference material.
- Galaxy Audio: not recommended. I have used some Galaxy Audio equipment that was usable but not high-quality.
- ART: not recommended. I have had problems with both ART products and their repair services.
- Behringer: not recommended. Behringer's products are designed to provide as many features as possible for the least cost.
- Nady: not recommended. I have seen major quality problems on Nady portable products.

# Chapter 14

# Measurements

Being able to take measurements is very important, as it removes much of the guesswork involved in making a sound system effective.

You should be able to do the following measurements:

- sound pressure level (SPL)
- realtime analyzer (RTA)
- (and others)

## 14.1 Hardware

Audio analyzers are available that can provide much of this functionality. However, the following hardware acts as additional audio inputs/outputs for a computer (via USB) and can be used not only for room measurement, but also for signal processing in general. These been selected for some combination of portability (size), versatility, and quality.

- Input: Shure X2U
- Output: Whirlwind pcUSB

You will also need a measurement microphone. I am using a dbx RTA-M, but there are probably better ones. Dayton Audio offers 3 test measurement microphones, and provides a unique calibration file for each microphone that they sell (cross-referenced by serial number). In the case of the UMM-6, you would not need the Shure X2U input adapter, but you also would be unable to connect a regular XRL microphone as input if you wanted to eg. try analyzing a microphone.

## 14.2 Software

- REW (Room EQ Wizard) is Java-based, and available for Windows, Mac OS X, and Linux. It provides many measurement functions. If you plan to use it, please consider reading all of the REW documentation.
- BRP-PACU is free and open source software for Mac OS X and Linux. It provides a pink noise generator and an analyzer and calculates transfer functions. It uses JACK for audio connections. See also the latest (as of 2015) blog entry regarding BRP-PACU.
- Jaaa, Japa, and Jnoise are all tools for Linux that connect to the JACK sound server (or standard ALSA) and provide signal/noise generation and spectrum analyzers.
- If you plan to do much signal processing on your computer, you may want to run the JACK audio connection kit on Linux. JACK provides audio routing between sound hardware and applications that support JACK. There are many applications that support JACK, and LV2, LADSPA, and DSSI plugins can be connected too.

## 14.3 Other things of interest

- Android apps from Keuwlsoft; especially SPL Meter (direct APK) and Spectrum Analyzer (direct APK). Measurement quality with your phone/tablet microphone is undefined. For good results, these could perhaps be used with a Dayton Audio iMM-6.

## 14.4 Todo

- http://johnr.hifizine.com/2013/03/room-eq-wizard-spl-calibration-without-an-slm/