Acoustic Voice Recording, "I am seeking recommendations for voice recording hardware…" REPRINTED WITH UPDATES

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This technical note presents information about acoustical voice recording for the voice clinic or voice training studio. It was originally published in "Perspectives on Voice and Voice Disorders" (2007). That paper is presented here in its entirety with a few additions at the end with new content (see copyright for permission statement). Content is written towards those who have little experience with audio equipment and can be used as a course supplement where basic audio equipment in the clinic or voice studio is discussed. Future updates will be available at http://www.ncvs.org/ncvs/library/tech.

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Keywords: Microphone, Voice Recording, Acoustic Analysis, Singing, Clinical Voice, Proximity Effect, Frequency Response, Directionality Plot.

1 Original Paper

In this section, a previously published manuscript presented.

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Title: Acoustic Voice Recording, "I am seeking recommendations for voice recording hardware…"

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(Note: The Denver Center for the Performing Arts location of the NCVS has been dissolved and is left here only for historical acknowledgments)

1.1 Introduction

At the National Center for Voice and Speech (NCVS) we are asked, often by new voice science faculty setting up a laboratory for the first time, to recommend equipment for voice acoustic measures. Questions also originate from graduates of our Summer Vocology Institute's Voice Instrumentation course who are eager to set up their own studio or laboratory. After receiving many similar requests for information, we wrote a short technical note describing the equipment needed for what we would consider a basic, although adequate, voice laboratory (Spielman et al., 2006). However, that technical note was a list of equipment with some general discussion, rather than a detailed tutorial on the reasoning behind selecting the right equipment, or how to use it. Further, recent postings on the ASHA Special Interest Division 3 (SID3) online forum indicate that individuals working in both clinical and non-clinical settings routinely encounter situations where recording and analysis equipment need to be purchased.

This paper was written to provide guidance in response to such questions. A deeper and more accurate understanding of equipment specifications can help guide purchases so that each particular location creates the recording and analysis environment that suits its needs. However, it is important to understand up front that without testing every piece of equipment, we cannot give recommendations for a specific equipment setup; nor can we provide acoustic analysis tutorials. Equipment and acoustic measures are by no means static. Nevertheless, we hope to offer as a general starting point the types of questions you should ask yourself before purchasing and using equipment, and provide information about some of the potential pitfalls associated with voice recording in clinical and research settings.

In this paper, we will first discuss the basic features of recording equipment and environments. Then we will demonstrate how the equipment you choose is affected by your selected analysis measures and recording environment, and offer two different examples from the NCVS: (1) a clinical setting, in which time, money, and equipment may be limited and the acoustic recording environment is typically less than ideal; and (2) a voice research setting, where time and money are budgeted for the setup and use of carefully selected equipment and recording environments to obtain consistent and reliable acoustic metrics.

1.2 The Basics of Recording

The primary components needed to record an acoustic voice signal are a microphone, microphone preamplifier (mic-pre or pre-amp), recording apparatus, and a room (in other words, the recording environment). Even if you are using a simple desktop tape recorder, these components, although somewhat "invisible", are still present. To record digitally, the setup also requires an analog-to-digital converter (AD), a digital interface, and software interface. Figure 1a illustrates each of these components; note that several components are contained in a few larger electronic bundles. Simpler and/or less expensive digital recording systems have the same components, many bundled inside the computer (Fig. 1.b). The more reliabilty you need in your sound data, the more you need to pay attention to each of the components in this chain.

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Figure 1 about here

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1.2.1 MICROPHONE

There are basically two types of microphones that can be used for recording: a unidirectional microphone and an omnidirectional microphone. When considering microphone selection, the first concern is the recording environment. Unidirectional microphones are more sensitive to sound coming from a single direction, and therefore may be preferred for noisier recording environments. Omnidirectional microphones, which usually provide a more accurate representation of sound, are sensitive to sounds from all directions and appropriate only in quiet rooms. Directional characteristics are typically shown in a microphone specifications sheet as a polar plot (Figure 2).

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Figure 2 about here
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A crucial distinction between these two microphones is not only their sensitivity to direction, but also their sensitivity to the audio spectrum. This characteristic is called *frequency response*. Generally, unidirectional microphones have more variable frequency responses, whereas omnidirectional microphones tend to treat all audio frequencies more evenly. Figure 3 shows a stylized frequency response similar to a common unidirectional voice microphone. Notice that nearly every frequency in the audible range is either attenuated or amplified, in other words, the loudness of the pitches in a sound are not equally captured. Other audio equipment will also have a frequency response but usually microphone and loudspeaker frequency responses are the most varying across the spectrum, thus most critical to know. All unidirectional microphones will also have what is called a *proximity effect*, an acoustical artifact which emphasizes lower frequencies when the microphone is close to the sound source (Figure 3). Both the frequency response and proximity effect may have a direct impact on how your signal is captured, and therefore on your analysis. In particular, analysis of intensity, spectral amplitudes, or even voice quality may be prone to these artifacts. It is therefore important to choose your microphone wisely, and compensate for known effects if needed. Because the extent of these effects varies with different microphones, and because it is not always detailed on the specification sheet, it is important to ask the manufacturer before selecting one for clinical research. Even then, specification sheets may show the frequency responses and proximity effects measured from a distance that is further than one would normally use (e.g., measured at 1 meter, for a head-mounted microphone), making the specification uninformative.
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Figure 3 about here
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1.2.2 RECORDING ENVIRONMENT (ROOM)

As mentioned above, equipment choice may be driven partly by the characteristics of the available recording environment (i.e., a quiet laboratory/studio versus a noisy, reverberant clinic room). In general, reliable data may be difficult to collect from an acoustically uncontrolled room (Fig. 4) because of extraneous noise as well as reverberation (sound reflections within the room). While an anechoic chamber is the most acoustically controlled environment, it is perceptually unsettling and physically uncomfortable for voice subjects. When recording research-level voice data, the most common choice is a high-quality sound isolation booth. An acceptable sound isolation booth should have heavy construction, double walls, a floating floor, quiet lighting and ventilation, and adequate patch bays for connecting signal sources to equipment outside the booth. Treat these booths with respect. Indiscriminate painting of the interior walls can seal off meshing to the acoustic insulation within the walls and get paint on the insulation, both of which effectively reduce the absorption and reverberation control.
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Figure 4 about here
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When a sound isolation booth is unavailable, treating a small room acoustically with commercial products (e.g., Auralex) can help alleviate some of the more common, minor problems; however, simply adding material to an otherwise noisy room does not solve fundamental acoustical problems. If reliable voice research is the goal, sound isolation and low reverberation are crucial.

In a clinical setting, you are often stuck with the room you have. The degree to which room characteristics can be controlled affects flexibility in terms of microphone selection (Fig. 4). Choosing the right microphone for collecting an acoustic signal in a particular environment should be approached in terms of type and specification. For example, clinical equipment or hallway noise may be minimized by using a unidirectional head-mounted microphone, which helps by (1) moving your microphone closer to the source, thereby reducing the noise level, and (2) picking up sound originating from a specific direction. However, as discussed above, these microphones have *proximity effects*, potentially affecting the accuracy of a voice recording. Nevertheless, fundamental frequency (F0) extraction and other frequency-related metrics should be impervious to all of these effects, though intensity and spectral based metrics will not. There is no bad microphone, but one must properly match a microphone to the intended environment and purpose.

1.2.3 PRE-AMPLIFIERS, A/D CONVERTERS, AND RECORDING APPARATUS

After an appropriate microphone is chosen, the rest of the signal chain leading into the recording device, including a mic-pre and an AD converter, must be selected. Like the microphone, these other components can affect the accuracy of a collected voice signal; thus, careful consideration must be given to the frequency response and the features present at all stages of signal transmission (Fig. 1). Be aware that lax standards of specification measurement and reporting exist in the pro-audio world, and specification sheets provided with equipment sometimes provide an incomplete picture. It is advisable to consult an acoustic expert when making the final choices for a recording setup. At this stage, equipment novices are often drawn towards high-end music stores (the ones that serves professional musicians in your area) to ask for advice. Remember, though, that those professionals have different goals, focused more on the musical quality of a recording than the accuracy of capturing a voice signal. Instead, consult an experienced acoustician or a sound engineer who has built a studio or laboratory, and a voice and speech scientist with expertise in this area.

The final stage in the signal chain is the actual storage of the audio information by a recording device. It should be noted that many recording devices serve as “all-in-one” packages and can include built-in microphones, mic-pre’s, and A/D converters. The storage medium itself is an important consideration. Recorder options include hard disk (both stand-alone and traditional computers), flash memory, direct to CD, DAT (digital audio tape), and Minidisc. Flash memory is used in compact devices intended for portability, while hard disk systems offer the most flexibility, particularly when using a traditional computer with recording software. Direct to CD and DAT systems can’t match the reliability of the first two types due to inherently fragile media; *they should be avoided*. One must also carefully take note of the file type used by a recorder. Mid-priced systems and portable devices often use compressed audio formats like MP3. These file types seriously alter the acoustical information being collected and should not be used. Make sure the recorder in question can record to uncompressed WAV files.

1.2.4 ADDITIONAL EQUIPMENT

If intensity measurements are desired, it is important to maintain a consistent microphone-to-mouth distance throughout the data collection session, making a head-mounted microphone ideal. A purchased or custom-made mount that is comfortable and secure allows subject flexibility while maintaining accuracy. It is important to *calibrate the intensity*, or reference to a known intensity level, measured from a microphone; to do this, a high-quality Sound Level Meter (SLM) is required. Unfortunately, the calibration of a microphone to the SLM is often done poorly. The most accurate (and detailed) calibration method we would recommend was illustrated by Svec et al. (2003). In the clinic, the alternative is to simply purchase an inexpensive SLM and read numbers directly from the display. This is perfectly acceptable for clinical work not intended for publication. Clinical systems that include calibrated microphones are also becoming more widely available.

In summary, when choosing equipment it is important to know your room and what you want to measure, and be aware of how each component contributes to the recording. A high-end system used in a poor environment may offer no better recording than a basic system in a good environment. Also, the old saying about “the weakest link” applies here as it does in most situations: choosing an outstanding microphone and coupling it with low-grade amplifiers will not yield outstanding results. So, don’t skimp on the cables!

1.3 Examples from the NCVS

At the NCVS, collection of voice signals for acoustic analysis is central to our institutional goals. Voice recordings may be taken for general voice diagnostic measures in our clinic, as well as for our voice research. In deciding on the appropriate equipment to use, we ask ourselves three questions: 1) What do we want to measure? 2) Where are we going to measure it? and 3) How much are we willing to pay, in time and equipment?

1.3.1 Example 1: Clinical Voice Recordings

The Center for Voice and Swallowing Services provides clinical voice care at the NCVS. The primary purpose of the clinic is voice diagnosis and rehabilitation. All recordings are collected in a clinic environment – small evaluation or therapy rooms with some reverberation and extraneous noise. The discussion below reflects our current thinking regarding the two remaining questions: What do we want to measure? and How much are we willing to pay?

The equipment we choose is first affected by why we’re collecting the voice signal. We may collect pre- and post-treatment samples to aid in evaluation and document change throughout treatment, or multiple samples to provide feedback to clients during therapy. We may also use acoustic measures to analyze voice signals for clinical research.

If our goal is simply to have pre-/post-treatment samples or to provide feedback during therapy, a good recording but not necessarily an extremely accurate recording is needed. We collect such samples with a decent recorder (tape) and a table-top microphone mounted on a stand, or more often a laptop computer with a head-mounted microphone and free or inexpensive recording software (e.g., WaveSurfer, Audacity, Goldwave). In either situation, we are careful to preserve a set mouth-to-microphone distance, recording environment (same room), consistent equipment, and recording input levels. These small details allow us to have control over our recordings and provide a more accurate picture of changes following therapy. For example, if a client increases vocal loudness but the microphone input gain or distance to the mouth are not consistent, this change may not be recorded.

Signal accuracy is more important if we plan to acoustically analyze the recording from our initial and/or final assessment. In this case, we add one more level of control, a small microphone preamplifier connected through the 'line-in' on a laptop soundcard (Fig. 1b, *right*). One should be aware that sound cards available in both PC's and laptops are likely adequate for most clinical applications, but we avoid using them in our clinical research as the internal components may be suspect (Ternstrom, 2007). PC sound cards and similar hardware can affect common voice acoustic metrics such as harmonics/noise, cycle-to-cycle variability (e.g., jitter and shimmer), and intensity. If we plan to use our recordings in clinical research, we use a setup similar to that in Fig. 1a; or, preferably, we go to the voice laboratory (see below).

When documenting disorders of the voice and evaluating change over time, we regularly use measures as simple as maximum phonation time, F0 range, and intensity; we collect these quickly and cheaply using a stopwatch, a laptop with a decent microphone and free software (e.g., WavesSurfer, PRATT), and an inexpensive SLM set at a prescribed distance (50 cm is typical). We also visualize computer recordings of sustained phonation using inexpensive spectrographic programs like VoceVista to inspect the phonation for noise, breaks, or frequency/intensity variability. We find it is often more beneficial in a clinical setting to carefully and consistently inspect a signal for disordered features than to run a numerical acoustic analysis such as jitter and shimmer. This is primarily because such analyses truly require high-quality microphones (Winholtz & Titze, 1997b; Winholtz & Titze, 1997a), high-quality AD conversion (Ternstrom, 2007), controlled recording environments, and signals without significant aperiodicity (Titze, 1995).

If more objective analysis is desired, it is crucial to know how the software might affect the output. For example, F0 is analyzed differently by different software packages. To address potential errors, we observe the output and identify and delete clearly aberrant points that would otherwise either inflate or diminish the mean F0 and standard deviation (this obviously takes some time, and also requires a reference for normative values). While the resultant mean, mode, and standard deviation of F0 is relatively robust for periodic voice, it is important to closely scrutinize a software package's minimum and maximum output before reporting these values in clinical histories or publications. Further, the microphone must be properly calibrated when recording a signal for intensity analysis. An improperly calibrated microphone will cause errors in intensity-based metrics; e.g., Voice Range Profile (Schutte and Seidner, 1993) or Dysphonia Severity Index (Wuyts, et al. 2000). Remember, when analyzing measures numerically, it is crucial to check those measures to make sure they are accurate enough to report.

To summarize, our clinical setups may consist of any of the following: 1) a medium-quality tape recorder with a table-mounted unidirectional microphone (Shure SM45); a stopwatch; an inexpensive sound level meter (Radio Shack); a tape measure; and a chromatic guitar tuner for measuring pitch (Korg) or an inexpensive keyboard; 2) an AKG head-mounted microphone; an inexpensive mic-pre (ART MicroMix); a laptop with built-in sound card; and VoceVista software; 3) a Kay Elemetrics CSL hardware and software system with accompanying AKG head-mounted microphone and computer. The first system is adequate for most voice evaluations and therapy but does not allow for detailed acoustic analysis of the voice. The second system records directly to computer and has the added feature of a separate mic-pre, allowing for more control over the microphone and recording. The third system can record at very high quality but may be too expensive and unnecessary for an acoustically uncontrolled clinical environment.

1.3.2 Example 2: Laboratory Voice Recordings

The voice research laboratories at the NCVS are engaged in multiple ongoing voice research protocols (e.g., efficacy-based Parkinson's Therapy, LSVT; Occupational Voice Loss; Vocal Economy Therapy; and Voice Forensics). The example given below will be discussed in light of the three questions listed earlier.

What do we want to measure? Our voice data are collected for scientific publication in peer-reviewed journals. Often, our goal is to find small changes in non-disordered voices, allowing for only a small degree of experimental error; this requires all aspects of the protocol (i.e., coaching of vocal tasks, recording equipment, equipment use, and recording environment) to be known, carefully controlled and consistent. Because we need high quality, accurate voice data, equipment expense is a secondary concern.

What room will we use to collect the selected measure? The NCVS at the University of Iowa site has access to an anechoic chamber; our main laboratory headquartered in Denver, where most of our acoustic research occurs, has several sound isolation booths large enough to accommodate a subject, a data collector and equipment. Sound isolation allows us to reduce artifacts such as reverberation or extraneous noise, and chose more sensitive recording equipment.

What equipment will give us the best recording for the environment? Since the isolation booths are controlled environments, we use a durable omnidirectional microphone mounted close to the mouth as our primary acoustic transducer. Our microphone (Countryman B3) has an extremely flat frequency response over the entire hearing range (20-20kHz +-3dB) and less than +-1 dB over the range of interest in our study. It uses a custom-built head mounting to avoid changes in distance, and it has no proximity effect. A Type I SLM (B&K 2238 Mediator) is mounted on a stand with a 30 cm string attached for quick and repeated measurements during initial calibration of the microphone. After the calibration procedure, the SLM is retracted and simply acts as a chatter channel in the final recording. We also use non-acoustic transducers (resprigraph bands and EGG) and a nontraditional acoustic transducer (accelerometer on the jugular notch).

The next block in the signal path after the transducers is the mic-pre. In traditional research protocols, a Kay Elemetrics CSL or similar system is adequate as the mic-pre. However, because the accelerometer signal is extremely low, we use a mic-pre with lower electronic noise (all electronic equipment adds some noise to a signal). The microphone, accelerometer and SLM, as well as the non-acoustic transducers, connect to a Millennia HV-3D 8 channel mic-pre (Millennia Media). There are several benefits, besides its high quality, that helped us choose this mic-pre: its industry-known low electrical noise, large range of amplification (8.0 dB - 61.5 dB), and stepped gain-control knob on each channel set to a very precise 1.5 dB. This last feature allows us to change gain mid-recording, without having to recalibrate (with adequate note taking for later software adjustment).

The pre-amp delivers the signal to an RME ADI-8 DS AD/DA Converter and an RME Multiface II 36 channel digital audio interface (RME, Germany). The digital interface to our primary recording computer is via a proprietary PCI expansion card on the motherboard. A type

II PCMCIA Cardbus interface is also available for use with a laptop when a more mobile recording unit is necessary.

Our primary recording PC has two hard drives, an OS drive (where Windows lives) and a data drive to which all recorded data are written. This allows for more efficient disk access during recording (Ternstrom, 2007). The digital interface is controlled by accompanying software and all audio files are collected using Cubase SE3 multitrack recording software (but any pro-audio software would be able to do this; e.g., DigiDesign ProTools or Apple Logic)

1.3.3 Miscellaneous Items

For completeness, it is important to discuss AC- and DC-coupled signals. DC-coupled signals capture static information down to 0 Hz, whereas AC-coupled cannot record information lacking variability (0 Hz). Virtually all pro-audio equipment is AC-coupled and has built-in filtering to prevent the capture of information below 20Hz. Such systems cannot be used to collect tidal flow, intraoral pressure, resprigraph traces, and other low-frequency signals. There are several options available. Our primary pre-amp is for only AC signals, but we have a 3-channel Frequency Modulator (custom built by Andrew Starr) which encodes DC input signals (onto a 10kHz carrier); demodulation is performed by in-house computer scripts. Other devices we have at our disposal include DATAQ, National Instruments, and several Kay Elemetrics CSL systems that have DC coupling as an option.

Data storage. In our current research protocol, a single subject recording session results in nearly 2 gigabytes of WAV files, sampled at 44.1kHz and 24bit resolution. At that rate, data storage becomes an issue. We have found that lossless audio compression is invaluable (free FLAC; http://flac.sourceforge.net/). Not only does FLAC allow for lossless compression of WAV files (note MP3 is not lossless compression), it has a built-in ‘check-sum’ that verifies the accuracy of compressed files after archiving (CD-R can have a poor shelf life and the ‘check-sum’ allows for verification of the archive).

1.4 Conclusion

The devices discussed above are by no means exhaustive, or even potentially the best quality or value for other recording environments or purposes. The items were chosen for our specific needs, environments, and experience in our own clinics and laboratories; there are many other fine manufacturers and quality hardware devices. Remember, an inexpensive piece of equipment may not be inexpensive to operate, nor is an expensive piece of equipment necessarily easier to use or more accurate than a cheaper one. Excellent references for both clinical work (Baken & Orlikoff, 2000) and efficacy research (Kent et al., 1999) offer more detailed discussions of some of the most important issues surrounding voice measurements. Further, new resources continue to appear in a wide variety of voice-related journals, conferences and websites as technique and technology changes; therefore, it is crucial for us as voice clinicians and researchers to remain current.

1.5 Figures

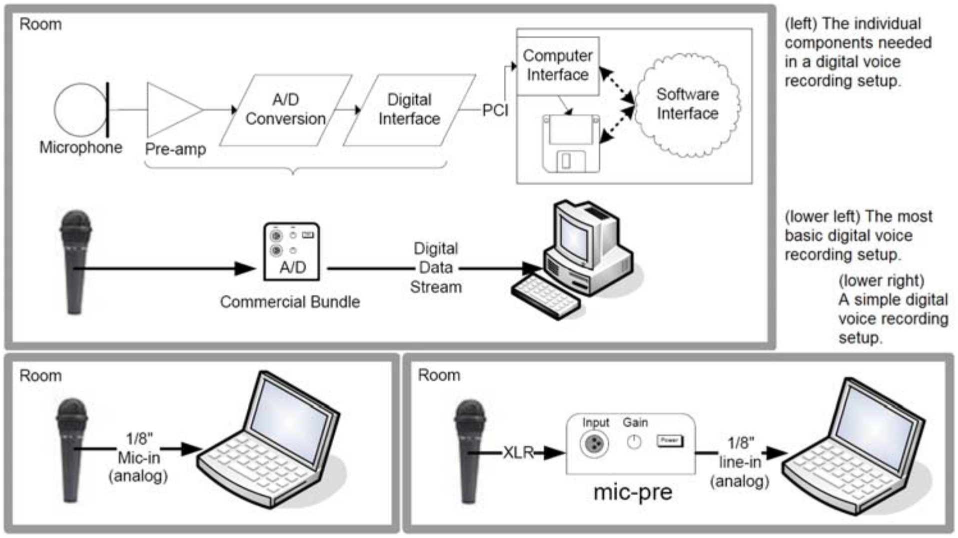

Figure 1. A digital voice recording setup requires a microphone, microphone preamplifier, analog-to-digital converter, digital interface, software interface, and recording environment. Many of these components are often bundled together.

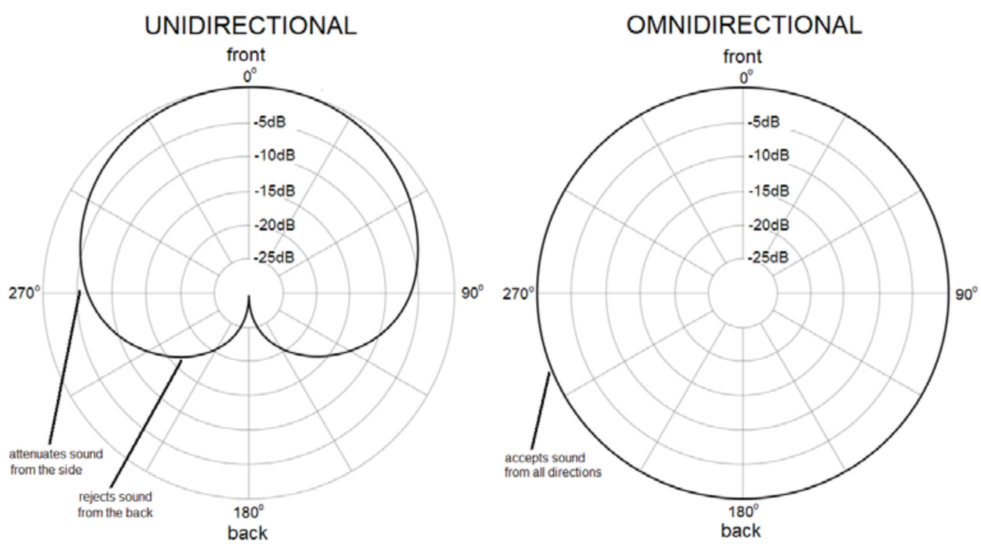

Figure 2. (left) *Unidirectional Microphone:* A microphone that is sensitive to a particular orientation or direction. This directional sensitivity is usually illustrated by a polar plot (360° circle with the front of the microphone at 0°) showing directionality in dB of attenuation (as opposed to amplification). This particular shape is called a 'cardioid', hence the common term 'cardioid microphone'. (right) *Omnidirectional Microphone:* A microphone that can pick up sounds from virtually any direction in an equal fashion, illustrated by a circle on the plot.

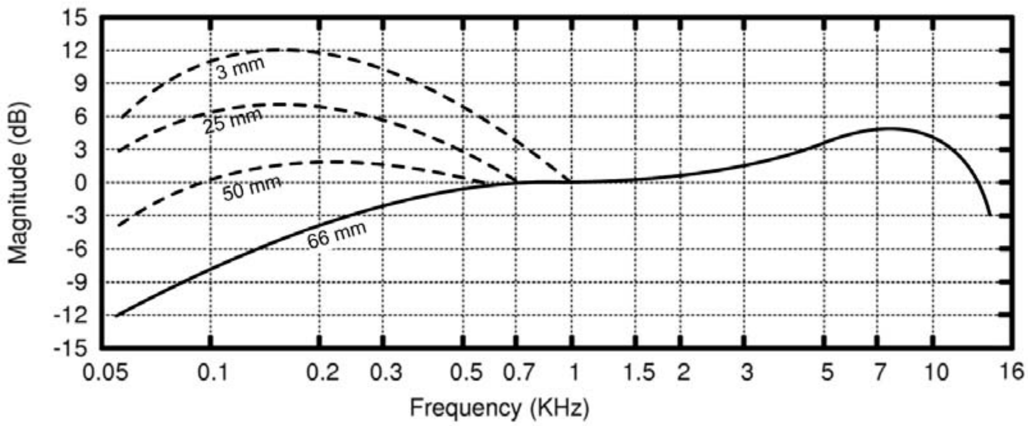

Figure 3. (solid line) *Frequency Response:* The sensitivity of a component to various frequencies (i.e., the amount in dB each frequency is amplified or attenuated where 0 dB means no change), usually graphed as an amplitude or gain (vertical axis) plotted against frequency (horizontal axis). (dashed lines) *Proximity Effect:* For directional microphones, the frequency response of the microphone changes at the low frequency end, as the microphone-to-mouth distance changes.

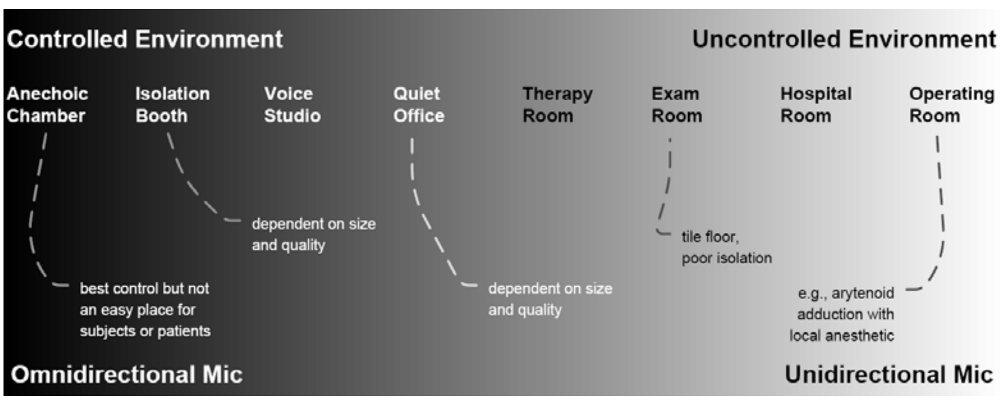

Figure 4. An example of various rooms where voice signals are recorded, illustrating the range from the most controlled to the most uncontrolled acoustic environment with preferred microphone type.

2 Additional Comments and Updates (by Eric Hunter)

Since the publishing of the above paper, audio recording options have continued to change. For example, more and more people are using their smart phones for recording and as inexpensive sound level meters (e.g. RTA Audio Analyzer, RadonSoft; FrequenSee, Daniel Bach; Rehearsal Assistant, urbanSTew; SPL Meter, By Studio Six Digital). However, the basic principles of microphone usage and recording environments have not. For example, while the smart phones have inexpensive or free sound level meter applications, the user does not know the quality of the microphone or the analog to digital hardware. The microphones in smartphones are not very good so these options should be primarily for approximations.

For inexpensive sound level meters, a standalone device is more stable than smart phone options and can, at a minimum, be compared to other sound level meters. There are many very inexpensive heap meters available on Amazon for $20-$30. At the NCVS, we use these type mostly for educational work but they would be more than adequate in a clinic. However, there is no audio out on many of these, a feature you would get with an inexpensive Radio Shack sound level meter.

Simple recording options in a clinic and voice studio can be accomplished with a stand alone recorder. These come with various ranges of microphones and features—usually recording directly to SD card. They range from just under $100 to several hundred. At the NCVS, we an inexpensive (low feature) TASCAM DR-08 and several Roland R-05. These are very convenient devices and the SD card can be read by most laptops with the audio files put into medical records or taken home for practices.

At the NCVS, we have also been using a new piece of recording hardware sold by ADInstruments (PowerLab 16/35). While this is not necessarily made for audio specifically, a microphone can be used as one of the acquisition devices. Additionally, many other types of devices can also be fed to the acquisition system, including EGG, accelerometers, pressure/flow systems, and respiration bands. The system accepts DC signals and allows for virtual channels so real time math and logic can be added to the recording. However, this system is truly a research oriented system and is not cheap.

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This technical note also contains the final accepted manuscript which was published in “Perspectives on Voice and Voice Disorders” in 2007. Permission to use the manuscript in the final accepted (not published) manuscript was given on April 4, 2011 from Jean C. White, Associate Director for Communications, Special Interest Groups and International Liaison Programs, American Speech-Language-Hearing Association. The following is the permission: “we grant you permission to post on an institutional repository or your personal Web site the final accepted manuscript with full citation for the November 2007 article and a link to the published article.” The full citation is above and given here:

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http://div3perspectives.asha.org/content/17/3/7.full.pdf

Revisions

1.0 Eric Hunter: Primary content document (November 2007), formatted as a technical note and additional information (June, 2012).
1.1 Eric Hunter: small changes to make it more concurrent with authors locations (August, 2012).