Procedures for Audio Recording and Speech Sampling

Consider your purposes for obtaining a speech sample.

- What type of phonological analysis do you intend to accomplish?
- What kind of speech sample is needed for the analysis?
- Will you be able to transcribe the speaker while you obtain the speech sample, or will you need to record the sample for later transcription?
- Does the speaker present any behavioral or linguistic difficulties that will affect the quality or representativeness of the sample?
- What stimulus materials will you need?
- What technical aspects of recording will you need to know about prior to and during the recording?

This barrage of questions was to get your attention! Consider all the information that follows—then be sure to prepare and practice everything needed to obtain speech samples that are appropriate and efficient for phonetic transcription. The next section provides some background on technical issues in audio recording, followed by a step-by-step list.

SOME TECHNICAL INFORMATION ON AUDIO RECORDING

It is helpful to think of a speech signal as having three potential forms: an airborne acoustic signal that is produced by the talker and that rapidly fades as its energy is lost, an analog signal that is stored on magnetic tape (such as that used in conventional cassette and open-reel analog tape recorders), or a digital signal that is stored on magnetic tape, a diskette, a compact disc (CD), or a Mini-Disc. When we analyze speech, we generally work from one of these three forms. Because the actual acoustic signal of speech (the airborne signal) vanishes as soon as it is produced, we need to store the signal if we are to listen to it again or to analyze it with acoustic techniques. The acoustic signal is an analog signal, meaning that its amplitude varies continuously with time. A microphone is used to convert this acoustic signal to an electrical signal that is then fed to a storage device. An analog magnetic recording, such as that done with ordinary (nondigital) cassette recorders, stores the signal as variations in a magnetic field. The magnetic tape recorder is a device that converts the electrical energy from a microphone to magnetic energy. As the tape passes by the record heads of the tape recorder, the metallic particles in the tape retain a magnetic field that represents a signal. When we want to play this signal, the reproduce heads convert the magnetic field to an electrical signal. The whole process depends critically on the movement of magnetic tape past the record and reproduce heads of a tape recorder.

Recordings of speech go back more than 100 years, beginning in 1877 with Thomas Edison, who recorded his voice on a cylinder phonograph with the intention of developing a dictation machine. The first flat disc recording was made by Emile Berliner in 1887, and the first magnetic recorder was invented in 1898 by Valdemar Poulsen, who used a steel wire as the recording medium. The technology of digital audiotape (DAT) was first used in recording studios in the late 1970s, and the CD became popular in the early 1980s. Because of the shift in recording technology to digital methods, it is important to understand some of the basics about digital signal processing (DSP).

A digital recording leaves the analog world and is stored as a series of numbers (hence, digits) in a digital computer or in a digital storage of some kind (e.g., DAT, CD, Mini-Disc). The process of digitization is discussed later in this appendix (but for extended discussion, see Baken and Orlikoff, 2000; Kent and Read, 2002). For present purposes, we note simply that a digital computer stores numbers; if we want to store a speech waveform in a computer, we must convert the analog waveform to a series of numbers that represents the waveform. The digital information can be stored on a magnetic tape (called a digital magnetic tape) or on a disk. When we want to hear this stored signal, we rely on a process that converts the stored digital information to an analog signal.

Therefore, the use of digital technology for storing a speech signal requires two kinds of conversion: Analog to digital (A/D) conversion and digital to analog (D/A) conversion. The three different forms of the speech signal are essentially...
interchangeable in that a given airborne acoustic signal can be recreated from either an analog or digital recording.

The goal in high-fidelity recording is to make a stored version of the signal that is as faithful as possible to its original production. To accomplish this goal, it is necessary to use proper equipment and techniques to minimize noise and distortion. The discussion that follows is organized to consider (a) the physical environment in which a recording is made, (b) the recording microphone, (c) the recording device (either analog or digital), and (d) recording procedures and pitfalls.

**Arranging the Recording Environment**

**General Considerations.** A first step that is all too often neglected is to evaluate the physical setting in which a recording will be made. A few small efforts can greatly enhance the quality of recordings. The primary goal is to determine if there are any obstacles to successful recording and, if possible, to remedy those that are found. A common problem is background noise that may contaminate speech recordings. Unfortunately, many recordings are diminished in value because of excessive noise on the tape that could have been prevented by a few easy steps. Some sources of noise include heating and air conditioning systems, fluorescent lighting, paging systems and telephones, bathrooms, corridors, elevators, playgrounds or parks, heavily traveled roadways, and electronic equipment (including personal computers that may generate fan noise). One source of noise that is sometimes overlooked is the recorder itself, especially when an internal microphone is used. If a source of noise is discovered and cannot be eliminated, it may be necessary to choose a different location (such as another room or building) for the recording or to record at times when the noise is less likely to occur (such as early mornings or evenings). Sometimes, it helps simply to move the recording equipment to a different position in the room (for example, away from a fan in a ventilating unit). As discussed later, the choice of microphone also can be important in reducing noise in a recording situation.

**Reverberations (Reflections).** A second potential problem relating to room acoustics is reverberation, or sound reflected within the recording room (Bachety, 1998). Reverberation is most likely to occur in rooms that have hard parallel surfaces (a typical situation in most buildings), on which sound waves can bounce back and forth. The result is a condition known as slap echo, which is especially disruptive for high frequencies. Use of drapes or other wall treatments can reduce reverberation. Another reflection problem is called near-field reflections. This problem occurs when a recording microphone is located close to a hard surface, such as a wall. Therefore, it is often better to place a microphone near the center of a room rather than close to one of its walls.

**Recording Equipment: Microphones**

There are many types of microphones from which to choose. In addition to cost issues, your choice of a microphone for phonetic transcription purposes depends on the type of recording you intend to make. Pressure zone microphones (PZM) and omnidirectional microphones are similar in that they can sense sounds produced at various locations in a room. A typical application might be a small room in which a child is free to move about, when the goal is to record what the child says wherever he or she may be. Unidirectional microphones with cardioid or hypercardioid pickup are designed for recordings of near sound and are therefore ideal for applications in which a particular sound target is of interest, such as one talker in a group. Parabolic microphones not only can record at large distances but also can pick up particular sounds if desired.

The technical terms for the different recording situations described earlier are ambient room recording, referring to recordings done in a general room environment, and close recording, referring to a microphone placement in close proximity to the person being recorded. Both types of recording may be of interest in a particular site. For example, ambient room recording may be preferred when the goal is to record the vocalizations from several children who are playing together, but a close recording is needed when the objective is to record one child only and there is little interest in surrounding acoustic events. To achieve these two different goals, it would be best to use two different microphones.

One common choice of microphone in the clinic is a full-size microphone that is mounted on a wall or ceiling or positioned on a microphone stand on a table top. Although this kind of microphone serves well for many purposes, there are some potential problems, especially when young children are being recorded. A common problem with both placements is that children may turn their heads so that the mouth-to-microphone distance is highly variable. Consequently, the strength of the signal will wax and wane with head movements. Although this variation in signal strength is not damaging for all applications, it can present serious obstacles to analyses that require close attention to sound patterns. Another problem that may occur when a table-mounted microphone is used is that children may bump against the table or pound it with their hands. Obviously, such noises will interfere with speech recordings.

The problems just described can be avoided by using specialized microphones. A lavaliere microphone, which attaches to a talker’s clothing, is well suited to applications in which one person is being recorded closely and there is relatively little movement. Lavaliere microphones do not always work well with young children who are highly active because body movements create motion noises that are readily picked up by the microphone. Another specialized microphone that works well in the clinic is a miniature head-mounted condenser microphone. Modern microphones of
this type can ensure high-quality recordings even when the talker changes head or body position, because the microphone follows head movement. Winholtz and Titze (1997) described a microphone of this type that is well suited to general recording needs. These microphones and their head mounts are light and usually can be worn comfortably for long periods. Of course, not all children will be eager to wear a contraption of any kind on their heads, but the enterprising clinician can often make a game of it to encourage cooperation. Finally, many general-purpose tape recorders have their own built-in microphones that vary widely in quality. It is wise to determine the characteristics of these microphones to be certain that they are suitable for a particular application. Although they are easy to use, internal microphones (located within the recorder) may not provide a signal of the desired quality. A particular problem is that they readily pick up noise produced by the recorder itself. In general, it is better to avoid using an internal microphone.

Recording Equipment: Analog and Digital Devices

The microphone must be connected to some kind of recording equipment. Among the choices are: (1) analog tape recorders with either reel-to-reel tapes or cassette tapes, (2) a DAT recorder, (3) a CD writer, (4) a digital disk device, or (5) a direct recording to computer disk. There are variations in quality within most of these choices, especially among analog tape recorders. As discussed earlier, analog recorders preserve the analog (continuous) nature of the signal to be recorded. In contrast, a digital recorder (DAT, CD, digital disk) stores a signal that has been converted to digital form. You are probably familiar with audiocassette recorders and CDs through your personal use. CDs offer large storage capability and good durability, making them popular for musical recordings. CDs function by recording a stream of data as tiny pulses on a plastic-coated aluminum disc. A laser beam then reads the data for playback. This method greatly reduces the potential for physical wear, which is another reason for the current popularity of CDs for musical recordings. For professional speech recordings, such as clinical recordings, a DAT recorder has become an increasingly popular choice. Because DAT recorders have controls that resemble those on analog tape recorders, most users adapt quickly to digital recording technology. We discuss next several factors that are essential to making a successful recording.

Choosing a Tape Recorder. When purchasing a recorder or choosing among recorders that may be available in an equipment room, users should keep in mind some minimum specifications regarding performance of the recorder (Baken and Orlikoff, 2000). The frequency response or frequency characteristic is a graph of the response of the system as a function of frequency. This graph shows the relationship between input and output of the system for different frequencies. A flat response is desirable, meaning that the response should not vary more than 3 dB over the frequency range of 30 to 15,000 Hz. Poor-quality recorders may have a limited frequency range and/or a highly variable frequency response. Signal-to-noise ratio (S/N) is the ratio of energy between the signal to be recorded and the background noise. Generally, the S/N should be at least 55 dB, meaning that the signal is 55 dB greater than the noise. Wow and flutter are variations in the speed of tape transport and should be less than 0.15 percent (unweighted) or 0.10 percent (weighted). Wow is a low-frequency variation, and flutter is a higher-frequency variation. Finally, the signal leakage between channels (left and right) should be less than or equal to 40 dB. If your recorder meets these specifications, it should be suitable for professional recordings.

Digital Recording (Digital Signal Processing). When a digital recording is made, it is important to know some basic information about digitization. Sampling rate relates to the frequency range available for recording and storing a signal. DAT recorders are available with different sampling rates (the rate at which the signal is sampled for conversion from analog to digital signal). The higher the sampling rate, the wider the frequency range (bandwidth) of recording. As a rule of thumb, the frequency range that can be represented is about half the sampling rate. Most DAT recorders offer a sampling rate of 44.1 kHz, which affords a highest frequency of about 20 kHz (excellent for most speech applications). For recordings of moderately high quality, sampling rates should be higher than 8 kHz. Bettagere and Fucci (1999) reported that listener-rated quality was superior for digitized speech sampled at 16 kHz than for analog tape-recorded speech. When a sampling rate of 8 kHz was used for digitized speech, the quality was essentially equal to that of analog tape-recorded speech. With modern equipment, there is very little reason to use a sampling rate of less than 16 kHz unless the objective is to record very long samples of speech, such as conversation. Especially when recording the speech of young children, a sampling rate of less than 25 kHz will very likely exclude the higher-frequency energy in fricatives (Kent and Read, 2002).

The other consideration in A/D conversion is quantization, which is expressed in bits. Quantization is the representation of signal amplitude, which changes in time during speech. The idea is to quantize (break into small pieces or “quanta”) the amplitude dimension so that changing values of amplitude can be represented as a series of numbers. Most modern DAT recorders offer 16- or 32-bit conversion. A 16-bit conversion permits 65,536 levels of amplitude to be represented in the digitized speech sample. This means that the amplitude variations that occur in a speech sample can be stored with up to 65,536 different numbers. Such a level of quantization ensures a high-quality signal free of the quantization noise that can occur with lower levels of conversion.
In short, the important thing to remember is that a 16-bit conversion should be used at minimum.

**Magnetic Tapes.** Recording tape consists of a thin (5 to 50 microns) plastic base that is coated with fine magnetic particles in a binder that allows these particles to change their orientation (and thereby store a signal). The base is commonly made of a synthetic material such as acetate, polyester (under the trademark name Mylar), polye vinyl chloride (PVC), or certain polyesters (especially PET film, the current standard for magnetic tapes). A primary concern in tape manufacture is to make tapes that resist stretching. Although Mylar is much stronger than acetate, it tends to stretch when pulled. This stretching destroys information recorded on the tape. Acetate tape breaks before it stretches, so that it can be repaired by splicing, with good potential for preservation of the recorded signal. The more recent types, PVC and tensilized-type PET, have high tensile strength that resists stretching. Another difference is durability: Mylar, PVC, and PET tapes hold up quite well, but acetate tapes disintegrate after about 15 years (Cudahy, 1988; Strong and Plitnik, 1983).

Cassette tapes come in formats such as C-60 or C-120. The number after the “C” indicates the total playing time in minutes. A C-60 tape plays 30 minutes on each side. Open reel tapes are gauged in mils. A standard 7-inch reel of 1.5 mil tape plays for 30 minutes at a tape speed of 7.5 inches per second (ips). The same tape plays for twice as long at a tape speed of 3.75 ips and for half as long at a tape speed of 15 ips.

Choose a magnetic tape that is suited to individual applications. If low cost is the primary factor, quality of recording may be sacrificed. Inexpensive tapes are often thin and subject to deterioration with use. More expensive tapes generally have a better composition of metal particles that retain the magnetic field. Analog tape comes in different types, three of which are commonly used: Type I (normal bias, ferrie oxide), Type II (high bias, chromium dioxide), and Type IV (high bias, metal). Cost and quality increase across this series. Type I tape is suitable for routine purposes. Type II offers a better frequency response and is preferred when signal energy above 12 kHz is important (as in music or fricatives in the speech of young children). Type IV also affords a good frequency response but is the most expensive.

The design and manufacture of tapes are constantly improving. Newer technologies for digital tapes involve the production of metal-evaporated (ME) tape, which may provide better durability than metal particulate (MF) tapes.

**Recording Procedures and Pitfalls**

The following suggestions pertain to obtaining a recording of high quality with potential for long-term storage.

**Recording Level.** Generally, tape recorders allow the user to control the amplitude of the signal to be recorded with a knob or slide that is labeled **recording level** or **input level**. Better-quality recorders have a VU (volume units) needle or LCD lights that indicate the recording level. The VU meter was specifically designed so that engineers working at telephone or radio terminals could monitor the transmission of speech or musical programs (Beranek, 1988). The standard VU meter is a colored scale that is graduated in either volume units or in percentages. Volume units should not be confused with decibels, which are used on other types of meters (such as sound-level meters). The red part of the VU meter’s scale corresponds to positive VU values (0, +1, +2, +3) or to percentages greater than 100 percent and is generally avoided in making recordings. The remaining part of the scale (usually yellow or white) corresponds to negative VU values (–1, –2, –3) or percentages less than 100 percent. For most recordings with analog recorders, the VU meter should not peak higher than 0 VUs or 100 percent on the VU meter. When the signal strength is highly variable, occasional peaks into the “red” zone are acceptable, so long as the average reading is in the “yellow” zone. However, the “red” zone should be avoided entirely when recording with a DAT recorder to prevent distortion.

**Automatic Level Control (ALC).** Many recorders are equipped with this function, also called automatic gain control (AGC). In general, do not use this function, because it distorts the actual strength of recorded signals. For example, consider a situation in which recordings are being made of a child who only occasionally produces speech and often does so with explosive bursts of short duration. When the child is silent, the ALC automatically adjusts the input volume to a maximum level. But when the child produces a sudden exclamation, the ALC circuitry quickly reduces the input volume, which may cause a loss of signal. ALC does have some benefits (particularly when the objective is to record sounds in a noisy environment and there is no interest in actual sound levels), but these rarely apply to phonetic transcription. We recommend purchase of recorders with ALC only if this function can be turned on or off as needs dictate. Although ALC may be desirable in some circumstances, it is not a good idea to use it without careful consideration of its consequences for particular applications.

**Noise Reduction Systems.** Because tape recorders generate noise, there has been a long-term interest in finding ways to reduce noise inherent to the recording process. Of the various approaches that have been developed, the two most successful are Dolby and DBX. The discussion of these systems is necessarily quite technical, and they are described only briefly here.

Dolby is named after the engineer who invented this process. There are different types of Dolby, only two of which are discussed here. Dolby-A uses two signal paths, a linear amplifier and a differential network. The output of the differential network is added to the “straight-through” signal
for recording the signal and subtracted for reproducing the signal. The differential network divides the frequency spectrum into four bands and applies its action only for the band(s) where it is needed. Dolby-B is a simpler process and is commonly used to reduce background noise in analog cassette recorders, which typically operate with a low tape speed and a narrow recording track. Dolby-B is similar to Dolby-A in that it uses a main signal path and a side chain. The effect of the side chain is to record low-level, high-frequency signals at a higher level, thereby improving the S/N. DBX uses compression to improve the high-frequency S/N.

**Distortion and Ghosting.** As noted earlier, magnetic tape holds the signal of interest in a magnetic field. The primary goal in magnetic tape recording is to saturate the tape with the desired signal and to avoid underrecording and overrecording. The former produces more noise than signal, and though it may seem strange that one can have too much recording level, excessive levels can result in distortion of sound quality and **ghosting** (or print-through). Ghosting occurs when a strong magnetic field on a portion of the tape “ghosts over” (or “prints”) onto the underlying or overlying portion of the tape on the spool. In effect, another part of the tape receives an additional and unintended magnetic field because it is wound over a very strong field. What does this sound like? Frequently, you will hear the ghosting as background sounds or as a kind of double recording (“echoes”) in which recorded samples are heard twice. In conclusion, it is always wise to refer to the operating manual to determine the optimal adjustment of recording level.

**Tape Stretch.** Another problem with tapes is stretch, especially with Mylar tapes. A new tape is wound at high tension on its spool. The first time it is used, this tension is partially released. With rewinds and replays, a section of the tape can be stretched, potentially causing “dropouts” (a noticeable drop in signal strength) or a change in tone quality. Remember that the tape must pass by the heads of the tape recorder if we are to record or reproduce a signal. If the tape is stretched after a signal is recorded, the signal will be altered on playback. Although some degree of tape stretch is unavoidable with repeated use of a tape, there are some steps that can reduce it:

1. Before a tape is used to make a recording, it should be fully fast-forwarded and then rewound. The purpose is to set the tension at a lower level than that set by the manufacturer and also to set the tension in accord with the tape deck under use.
2. It is better to use tapes that are thicker. This may mean avoiding tapes with longer play times (90 minutes or more), because they are typically thinner. These problems are not as severe with DAT as with analog tapes, but all magnetic tapes are susceptible to some mechanical wear and tear.

3. Be gentle: Avoid abrupt, vigorous, and unnecessary rewinds and replays. Stretch is particularly severe on the leading portion of a tape, which is subject to large forces developed during rapid rewinds.

Note that a permanent acoustic archive would have to be based on other recording media that are not typically used in general-purpose speech recordings. However, for most purposes, one can get satisfactory long-term storage from recording on digital disks or CDs or directly on to the hard drive of the computer. The major drawback to the latter is that speech samples consume large amounts of computer memory, and few of us want to fill our computer memories with stored speech samples. Therefore, a reasonable means of long-term storage is to use diskettes or CDs. These offer still another advantage, as discussed next.

Although we might like to think that tape recordings last indefinitely, they do not. In general, recordings deteriorate with time, particularly so when stored in an environment with high temperature and humidity. Both analog and digital tapes are metal particle tapes that are subject to eventual deterioration (Speliotis and Peter, 1991). For most magnetic media, deterioration can be detected within 5 to 8 years after recordings are made (Leek, 1995). Although control of temperature and humidity will extend the accuracy of the recorded information, errors ultimately will contaminate the quality of the recorded data. For short-term applications, these degradations may not be of much concern, but anyone who hopes to build a library of magnetic tapes for long-term use should be aware that magnetic tapes are susceptible to deterioration.

**Summary**

Making a high-quality recording is more than just pushing a “record” button. It is worth taking some extra time to observe the basic principles discussed in this appendix. With proper attention to a few details, you can make a recording that is virtually free of noise and distortion.

**SPEECH SAMPLING PROCEDURES**

**Step-by-Step Procedures**

1. Based on all the considerations reviewed in the previous discussion, select the appropriate recorder.
2. Learn the basic features of the recorder.
   a. Where does the power cord plug into the recorder?
   b. Where does the remote microphone plug into the recorder?
   c. How do you set the recording speed?
   d. What knobs or buttons control loudness of recording and/or playback?
e. Does the recording mode require simultaneous depression of two buttons?

f. How does the pause control work?

g. How does the remote off/on switch on the microphone work?

3. As underscored previously, use the manual volume control, not the automatic volume control. Automatic volume controls will distort beginnings and endings of speech and include unwanted background noise.

4. To avoid recording any noise emanating from the machine during recording, carefully position the recorder on a different surface and as far as possible from the microphone.

5. Prior to recording, announce the speaker’s name and the complete date on the tape. It is also useful to record this information at the end of the tape or disc when you have finished recording. Be sure to write this information clearly on the media container and on the label.

6. Use the identification recording as an opportunity to test your setup. Do you hear any other noise on the recording? Are all the buttons and dials working correctly? Recheck all settings!

7. When you are ready to record the speaker, place the microphone no more than six to eight inches from the speaker’s lips. To avoid “popping” noises, angle the microphone to point at the speaker’s nose, rather than mouth. Adjust the volume control so that the speaker’s vowels cause the needle on the VU meter to peak just below the distortion area. The consonants should be sufficiently audible to discriminate subphonemic features, such as unaspirated and frictionalized stops. Volume levels between one-third and two-thirds of full scale usually yield the signal-to-noise ratios required for narrow or broad phonetic transcription. Be sure that your utterances can be heard easily upon playback, with the speaker’s voice somewhat louder than yours.

8. If you must record in a noisy environment, minimize negative effects by reducing the mouth-to-microphone distance when the noise is constant. Have the speaker repeat any words that might have been obscured by a transient noise.

Evoking the Speech Sample

Procedures to evaluate or “test” speech usually include specific directions about how to obtain responses from the speaker. Standardized administration procedures are central to the validity of data obtained from articulation tests and word-repetition tasks. Free speech sampling should also be accomplished using standardized procedures with each speaker. The following recommendations will help you obtain rich conversational speech samples from young children with phonological disorders.

1. Be casual about the presence of the microphone so that periodic adjustments of the volume level or the placement of the microphone do not disturb the speaker. Most children can be trained to respect equipment (e.g., to not touch the microphone). Explain why you need cooperation. Adults are often more self-conscious about being tapped than children, but children will accommodate to the tape recorder if you are matter of fact.

2. Use a variety of materials and introduce different topics as needed to keep the speaker talking and to obtain representative proportions of parts of speech, word shapes, and phonemes. Medial and final /dʒ/ do not regularly occur in spontaneous conversational speech, and, therefore, no special procedures are used to evoke them. (See Shriberg and Kwiatkowski, 1985, for procedures and findings using five types of continuous speech samples: free, story, routines, interview, and scripted.)

3. If you are obtaining a free speech sample from a speaker who is difficult to understand, plan to gloss (repeat what the speaker says using a natural conversational style) after each utterance. It is not necessary to gloss every word—just those that the speaker intended to say that will be difficult to understand from the recording. Be sure to allow the speaker opportunities to clarify utterances to increase intelligibility for later transcription.

4. Make notes on articulatory behaviors that may not be perceptible on the audio recording, such as lip rounding or unrounding, unreleased stops, fricative distortions, and any facial gestures that may accompany speech production.

5. Make summary notes on the speaker’s general health, motivation, and physical state (e.g., whether congested or irritable during the recording); these factors could affect the validity of the speech sample.

Some Transcription Alternatives

One final aspect of sampling and recording to consider is the level of phonetic detail you intend to transcribe. For example, if you will be transcribing prosody, the speech sample needs to be representative of natural speech, rather than a narrative. Following are six other transcription choices discussed in this text (see Chapter 7) and in Ohde and Sharf (1992). You may not be able to make decisions on each of these and other transcription alternatives until you have some familiarity with the speaker’s error pattern.

1. Aspiration [ʰ]. [*]. Aspirated and unaspirated stops are predictable in English and therefore can be omitted when a speaker’s use of stops is not in question.

2. Stop release [*]. The release of postvocalic stops is optional (i.e., in free variation). Thus, it is not necessary to transcribe the release unless it is exaggerated or unless no release occurs where expected (e.g., past [p æ s tʰ]).

3. Nasality [~]. Nasal assimilation is predictable in nasal
consonant contexts and may not need to be transcribed unless noticeably greater than normal.

4. **Duration [\(\cdot\)], [\(\cdot\cdot\)]**. Predictable vowel duration may be omitted, using duration symbols only when they may provide information on consonant deletion or devoicing.

5. **Devoicing [\( \cdot \)]**. Predictable devoicing of final obstruents or consonants preceded by voiceless consonants (e.g., /\( r^1 \) in try) may not need to be transcribed.

6. **Vowel neutralization**. The precise vowel used in unstressed syllables (e.g., [\( \theta \)] or [\( \epsilon \)] or [\( \iota \)]) may not be important to capture for phonological analysis.

## REFERENCES


