SHURE®
LEGENDARY PERFORMANCE™

BASICS

HOME RECORDING & PODCASTING

SHURE®

LEGENDARY PERFORMANCE™

WHAT LEGENDARY PERFORMANCE MEANS TO SHURE

We believe that the phrase legendary performance should not be used lightly. At Shure, we take its connotations seriously and use past accomplishments as a foundation and roadmap for the future.

We start by thinking about performance as it relates to our products, and we work hard to ensure that they remain the "gold standard" of quality, reliability, and durability.

Equally, we are always conscious of our performance as an industry leader. We are committed to developing products that will provide the same high quality and reliability tomorrow as they do today.

We also rate our performance in the context of our relationships. Enabling others to fulfill their potential drives us to provide the best service, support and training possible. In this respect, we like to share our knowledge freely.

Ultimately, our 84-year heritage has been built on a diverse and storied foundation of legendary performances, and all of our activities revolve around optimizing your performance.

SHURE KNOWS HOW

For almost every application there are specially designed and optimized Shure microphones. This guide provides you a basic overview on how microphones work, what you need to consider to select the best product for home recording and podcasting, how to set them up and use them in recording applications.

HOME RECORDING AND PODCASTING

Home recording and podcasting are increasing in popularity every day. The equipment used for these applications has become more sophisticated, practical, accessible, and affordable — and more and more people are getting involved with these types of audio projects.

The purpose of this guide is to help you, the home recordist or podcaster, capture better sound for your recording projects, whether they are:

- Monologues
- Round table discussions
- Interviews
- Music performances, or
- Creating audio tracks for videos

The guide takes a step-by-step approach to discussing principles, products and placements, as well as helping you get past some of the most common problems.

If you are new to home recording and podcasting or if you want to improve your recordings, this is a great place to start!

WHAT IS DIGITAL AUDIO

There are many rumors and stories about how and when the first audio recording was done. Nevertheless Thomas Edison is reckoned as the pioneer in this field: on December 6, 1877 he recited the first stanza of the poem “Mary had a little lamb” on his tinfoil phonograph.

Since then many different formats were created to record and playback audio, such as the gramophone disc, the compact cassette, DAT, the MiniDisc etc. In the meantime digital hard disc recording has become the standard.

Analogue Signals

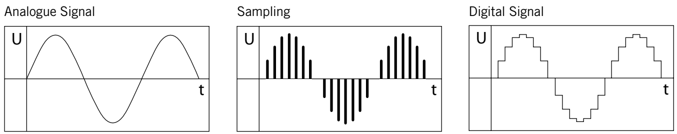
For all kinds of storing, the audio signal needs to be converted into an electrical signal. The electrical audio signal reflects the wave of the sound.

With an **analogue signal** a continuous voltage is conveyed. Such an electrical audio signal can easily be modulated onto a gramophone disc or a magnetic tape. A physical quality in a medium, such as the intensity of the magnetic field or the path of a record groove, is directly related, or analogous, to the physical properties of the original sound.

Digital Signals

Digital recording means converting an electrical analogue signal into a digital signal. The most important characteristics of a digital signal are the sampling rate and bit depth. In general, the conversion from analogue to digital involves taking periodic measurements or samples of the audio signal level and translating those measurements into a string of 0’s and 1’s. Expressed graphically this can be described as transferring a sine wave into a “stair” wave (see graph below).

The sampling rate describes how many times per second the analogue signal is measured. The higher the sampling rate the higher the possible maximum frequency response. A sampling rate of 44.1 kHz (the analogue signal is sampled 44,100 times per second) can accommodate audio frequencies as high as 22,050 Hertz, delivering “CD quality.” Lower sampling rates provide reduced sound quality (sometimes described as “speech quality“) but results in smaller file sizes and faster download speeds. Higher sampling rates are sometimes found on professional recording equipment, although there is debate as to whether sampling rates much higher than 44.1 kHz translate into audible improvements in sound quality.



The bit depth describes the number of digital bits used to store the measurement of the audio signal level each time it is sampled. Using more bits allows a more accurate measurement and a better quality recording, by increasing the dynamic range and reducing hiss. For example, an 8-bit sample allows the audio signal level to be measured in 256 discrete steps; if the actual signal level is somewhere between two steps, then the estimate won’t be accurate. A 16-bit sample (used on audio CD’s) allows 65,536 discrete steps, which is enough to create a very accurate estimate of the signal. Using more bits (e.g. 24-bit which are 16 million steps, or even 32-bit which are 4,300 billion steps) also results in larger file sizes and longer download times, however, this requires more processing power and memory when editing.

The resulting digital signal can be stored on different media such as CDs, DAT tapes of directly onto a hard disk.

CONNECTIONS

Before you can store a digital signal onto a hard disk, the analogue sound signal needs to be converted into a digital signal. In this chapter you will find various ways to connect a microphone to a computer.

Microphone

Balanced

In general, microphones provide an analogue audio signal. Professional microphones feature an XLR-output with three pins that transfer a **balanced** signal. One pin is ground, and the other two carry the audio signal. Pin 2 is the so called hot signal and pin 3 the cold. This method reduces the susceptibility of external noise while allowing the usage of longer cables.

Unbalanced

Entry level microphones often feature an attached cable with an unbalanced 6.3 mm or 3.5 mm connector. An unbalanced output carries the signal on a single conductor and is more susceptible to external noise. For that reason only balanced connections are used in professional miking applications.

USB

More and more professional USB microphones are available. A USB microphone is essentially a mic with a built-in USB audio interface that converts the analogue signal into a digital signal. It can be directly plugged into a computer without requiring an external audio interface.

USB microphones, such as the Shure PG27USB and the PG42USB with plug and play functionality, are an easy start into home recording and podcasting.

But how can microphones without a digital output be connected to a computer?

Sound Card

To be able to record analogue audio onto a computer hard drive, you need a sound card. In general, all laptops and most computers are equipped with one. Many internal sound cards are not shielded from electrical noise that is often caused by fans, hard drives, and the computer's own circuitry, which means noise and hum can be introduced into your audio. Also, most internal sound cards are not equipped with professional microphone connectors, do not provide phantom power for condenser microphones, and do not provide enough amplification or gain when working with low-level tracks or sounds.

For those reasons we recommend the use of a higher quality sound card that features balanced XLR inputs and an adequate microphone pre-amp. Such sound cards allow the direct connection of a professional microphone and high quality recordings with low noise levels.

High quality PCI cards are available for desktop computers. For laptops external sound cards with a digital output (USB of Firewire) are the best choice. Such cards can be connected to a computer easily and have one or more XLR inputs for professional microphones as well as a microphone pre-amp.

A quick and easy solution for all computers is the Shure X2u-to-USB adapter which features an XLR input and a USB output. The XLR input is balanced and the digital output is provided to the USB port of the computer. The X2u is compatible with PCs and MACs and can be operated without the installation of additional drivers.

INFO: Latency

Latency is a delay in the signal path caused by the time required to convert sound from analogue to digital (or vice versa) for processing of the signal.

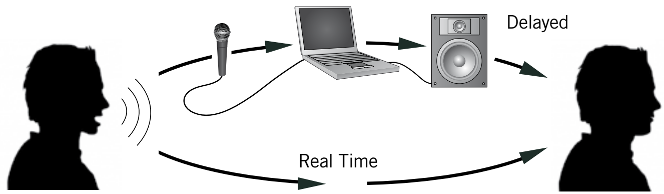
Usually measured in milliseconds, latency can occur at multiple points in the signal path… and it can really add up. What this means is that the sound you are hearing when you listen to yourself singing might not be happening in real time.

A latency of approximately 4 ms is normally uncritical. With a higher latency the timing of the musicians suffer.

E.g. if a singer is supposed to sing to a bandmix and the latency is too high, he will start singing too late.

Standard internal soundcards usually have a pretty high latency (200 ms and more). While this latency is not critical for podcasting and voice over, it is unacceptable for musicians recording a multi-track song.

High-quality sound cards offer a latency of 4 ms and less.



Mixers

A third possibility is using mixers (or “mixing console” or “mixing board”). Small mixers that offer a few balanced XLR inputs are sufficient for home recording. The output of such mixers offers a gain level that is high enough to be connected to the line input of the computer. Internal PC sound cards normally have an acceptable sound quality at the line input to be used in combination with a mixer. But as the mixers often have balanced XLR outputs, and the sound cards unbalanced inputs, special cables or adapters are necessary.

An advantage of a mixer is the option to blend separate microphone signals into one mix. Even though you would normally use a single track for each signal – instrument or voice – a mixer can help to make a quick and easy recording onto one track.

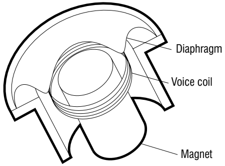
MICROPHONE BASICS

The transducer is the heart of the microphone. It converts sound into an electrical signal. The two most common transducer types are Dynamic and Condenser:

Transducer Type

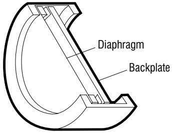
Dynamic

Dynamic microphones employ a diaphragm, a voice coil and a magnet. The voice coil is surrounded by a magnetic field and is attached to the rear of the diaphragm. The motion of the voice coil in this magnetic field generates the electrical signal corresponding to the picked up sound. Dynamic microphones have a relatively simple construction and are therefore economical and rugged. They can handle extremely high sound pressure levels and are largely unaffected by extreme temperatures or humidity.



Condenser

Condenser microphones are based on an electrically-charged diaphragm/ backplate assembly which forms a soundsensitive capacitor. When the diaphragm is set in motion through sound, the space between the diaphragm and the backplate is changing, and therefore the capacity of the capacitor. This variation in spacing produces the electrical signal.



Condensers are more sensitive and can provide a smoother, more natural sound, particularly at higher frequencies.

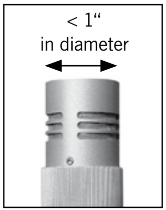
All condenser microphones need to be powered: either by batteries in the microphone, by phantom power provided by a mixer, a sound card or an external analogue to digital converter.

There are two main types of condenser microphones:

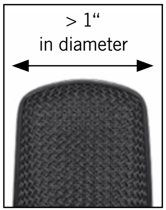
Small diaphragm – generally used for live performance and recording. They are called small diaphragm because the transducer’s diaphragm is less than one inch in diameter. Small diaphragm microphones provide a more natural sound reproduction and are preferably used for miking instruments.

Large diaphragm – traditionally favored by recording studio engineers and broadcast announcers, condenser microphones with a large diaphragm (one inch in diameter or larger) usually have higher output, less self-noise (the “hiss“ the microphone might make), and better low-frequency response, which can result in a “higher fidelity“ sound for both vocals and instruments.

Small diaphragm



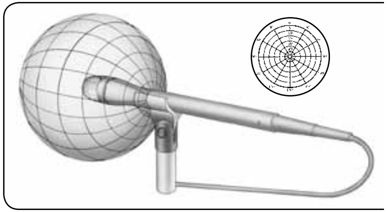
Large diaphragm



Pickup Patterns

Microphones are available with various pickup (or polar) patterns. The pickup pattern is the representation of the microphone's directionality. In other words, the pickup pattern describes the microphone's sensitivity to sounds arriving from different directions. The most important pickup patterns are:

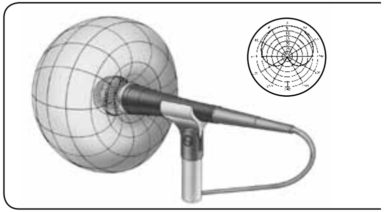
Omnidirectional



An omnidirectional microphone picks up sounds equally from all directions and reproduces the sound source more natural than a unidirectional microphone. It is good for natural room sound and group vocals. Also good for when the singer or talker may move around different sides of the microphone (but their distance to the mic stays the same).

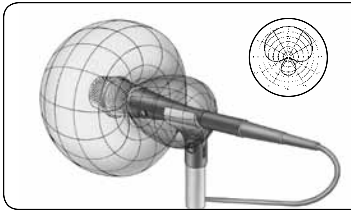
As an omnidirectional microphone picks up all the ambient sound in a room, e.g. the computer fan, it is not the recommended choice for home recording. However, if the goal is to enable listeners to hear what is occurring in the background, you should consider an omnidirectional pick-up pattern.

Cardioid



This is the most common type of microphone. It is called “cardioid“ due to its heart-shaped pick up pattern and has the most sensitivity at the front and is least sensitive at the back. This microphone helps reduce pickup of background noise or bleed from nearby sound sources.

Supercardioid



A supercardioid microphone is even more directional than the cardioid. Supercardioids have the tightest pickup pattern, further isolating the sound source. But they also have some pickup at the rear. Good for noisy, crowded spaces and when multiple microphones are being used, such as for round-table discussions where you want to keep the voices distinct.

INFO: Proximity Effect

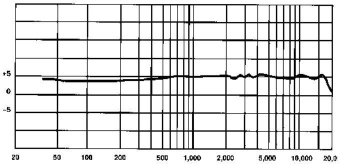
Every directional microphone (i.e. cardioid, supercardioid) has the so-called proximity effect. This is created when the microphone moves closer to the sound source resulting in an increase in bass response and, hence, warmer sound.

Most notably when one is recording vocals the speaker or singer needs to pay attention to the distance of the mic as every change in the distance creates a difference in sound.

Frequency Response

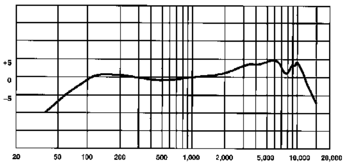
The frequency response is the output level or sensitivity of a microphone over its operating range from lowest to highest frequencies. Generally two types exist:

Flat Frequency Response



All audible frequencies (20 Hz – 20 kHz) have the same output level. This is most suitable for applications where the sound source has to be reproduced without changing or “coloring” the original sound.

Tailored Frequency Response



A tailored response has varying output levels across the frequency range and is usually designed to enhance a sound source in a particular application. For instance, a bass drum microphone does not need to reproduce high frequencies above 6 kHz or a vocal microphone may have a peak in the 2 – 4 kHz range to increase intelligibility.

Microphone Choice

Depending on the instrument or the voice, different microphone options are available. Many models are specialized for certain applications. Home recordists mostly do not have the budget to purchase different microphones for several instruments and vocals, but should consider buying one or a few models that are suitable for diverse applications.

Here some ground rules:

- For loud instruments, such as amplified guitars or drums, dynamic microphones are preferable.
- For acoustic instruments, such as guitars or voice, condenser microphones provide a more natural, detailed sound.
- If you want to achieve a warmer sound, dynamic microphones are the better choice – whereas condensers in general are more sensitive and more responsive to higher frequencies.
- Due to these opposite characteristics, both types of microphones (dynamic and condenser) should be part of the standard recording equipment or home studio.

The choice of microphone is essential to get the sound right when performing or recording. Equally essential is the placement technique which is described in the next chapter.

MICROPHONE PLACEMENT

Both the choice of the microphone as well as the placement technique can be optimized with some testing and experience. The following recommendations and placements are intended to help the beginner to get good results quickly. These tips are not written in stone but are mostly a matter of personal taste. So try different positions to see which sounds best to your ears. One thing that is true for all recordings is: the better the microphone placement, the easier the post production.

Why is the microphone placed upside down?

During vocal recordings the microphone is often placed upside down. This has more historical reasons. In the past tube microphones were very common in studios. The tube pre-amp omits heat and if it wouldn´t be placed upside down, the warm air would pass the diaphragm and increase the self-noise.

Vocals

A condenser microphone is the preferred choice for vocals as it delivers a more natural and nuanced reproduction of the voice. The standard choice for vocals is a large diaphragm microphone. It usually has a higher output, less self-noise and better low-frequency response, which can result in a “higher fidelity” sound.

Explosive breath sounds (which most often occur with “p“, “t“, “d“ and “b“ sounds) can be very distracting on the final recording. Windscreens and pop filters reduce these sounds. They also help to keep a consistent distance to the mic throughout the recording. Find more information on pop filters in section 10.

Keep the microphone approximately 20 to 30 cm from your mouth. This distance is close enough to minimize pickup of unwanted room reflections and reverberation, but far enough away to minimize picking up mouth and breathing noises. Do not get too close, though. “Eating the microphone“can decrease intelligibility.

Be aware that even a piece of paper can reflect sound and these reflections degrade the sound quality. So if you need to read lyrics or copy from it, make sure the sound reflections do not bleed into the microphone.

Spoken Word

Spoken word recordings, especially high quality podcasting, follow the same rules as vocal recordings. Simpler recordings, e.g. voice-overs for private holiday videos, can be done with dynamic microphones. The sound will be not as natural and nuanced but on the other hand breath sounds are reduced.

TIP: Intelligibility

Achieving the most natural sound reproduction is not necessarily the highest priority with vocal recordings - much more important is intelligibility.

Boosting frequencies in the 2 – 4 kHz range helps increasing the intelligibility, without making the track louder. Low frequencies make the voice sound warmer but they also diffuse the sound, so decreasing them is also a good option to improve the sound.

Instruments

Drums

The drum kit is one of the most complicated instruments to record. Although there are many different methods, some common techniques and principles should be understood. Since the different parts of the drum kit have widely varying sound they should be considered as individual instruments, or at least a small group of instrument types: Bass, Snare, Toms, Cymbals, and Percussion. During the microphone placement you need to not only consider finding the “sweet spot” but also to avoid picking up the other instruments at the same time. Supercardioid microphones are ideal for this application.

The **bass drum’s** energy is primarily focused in two areas: very low-end timbre and “attack”. Typically the microphone is placed inside the bass drum with a boom arm a few inches away from the beater head. The closer the microphone is placed to the beater head the more attack you will get, and therefore the further away the less. Placing the mic closer to the edge makes the sound less bassy but more differentiated.

Specialized microphones are recommended for miking a bass drum. Their frequency response and the big size are optimized for the low frequencies of a bass drum. They have a good response in the low frequencies, a boost in the attack range and above 6 kHz (approximately) frequencies are cut off substantially.

The **snare drum** is typically miked on the top head at the edge of the drum, approximately 2 – 7 cm above the drum head with a dynamic cardioid or supercardioid microphone. If you are using a supercardioid microphone be aware not to pickup the hi-hat from the back.

Toms can be miked with dynamic or condenser microphones in the same way as the snare drum. An advantage of small condenser microphones with goosenecks is that they can be directly attached to the rim of the tom and no separate stands are needed. Additionally condenser mics usually provide more attack.

Overhead microphones are typically a stereo pair. Not only to record a stereo image but also to capture all the cymbals with a consistent level. Flat frequency response condenser microphones will capture an accurate reproduction of the high frequencies. The mics are placed above the drum kit aiming to the cymbals. The closer they are moved towards the cymbals the less is picked up from the rest of the drum kit.

Many times the overhead mics will provide enough response to the **hi-hat** to eliminate the need for a separate hi-hat microphone. If necessary, a small diaphragm condenser mic placed away from the puff of air that happens when hi-hats close and within four inches to the cymbals should be a good starting point.

Simpler methods of drum miking are used for jazz and any application where open, natural kit sounds are desired. Using fewer mics over sections of the drums is common. Also, one high quality mic placed at a distance facing the whole kit may capture the sounds of kit and room acoustics in an enjoyable balance. Additional mics may be added to reinforce certain parts of the kit that are used more frequently (typically snare and bass drum).

Acoustic Guitar

Typically, a small diaphragm condenser microphone is used to mike an acoustic guitar – nevertheless a large diaphragm mic can also provide very good results.

Aiming the microphone directly at the sound hole creates a bassy, boomy and full sound. Is the mic too close to the sound hole, hand movements are audible and therefore an adequate distance should be created.

Moving the mic towards the neck adds harmonics and brightness but also fret noise. Depending on personal taste this may be desirable but it is not very common.

A balanced sound provides the placement of the microphone between the sound hole and the tailpiece.

To achieve an open, well-balanced and natural sound two microphones are needed. One is aimed at the body and the other one is aimed towards the neck (at the 12th fret). The microphone aimed at the body can be a large diaphragm mic to best reproduce the low frequencies. Depending on the room acoustics this microphone can also be placed one meter away from the guitar.

Electric Guitar/Electric-Bass

Even though miking an electric guitar or bass does not mean placing the mic directly at the instrument but at the speaker, still it is defined as picking up the instrument.

Before placing the mic at the cabinet you should make sure you know exactly where the speakers are located. Very often two or four speakers are mounted in a cabinet. Moving the mic towards the edge of the speaker cone results in a duller sound. At the centre of the speaker cone the sound is more balanced and has most "bite".

The proximity effect increases the low frequencies when a unidirectional microphone is placed very close to the speaker. Placing the mic further away reduces the bass and the mic picks up more room ambience.

A critical point with miking electric guitars and basses is the high sound pressure level and therefore dynamic microphones are mainly used for this application. If you prefer a condenser microphone make sure it is capable of handling the high volume.

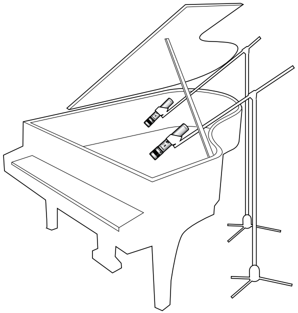
Piano

Picking up the sound of a piano is very challenging. On the one hand it is acoustically the widest instrument – the frequencies range from 27.5 to 4,200 Hz; with harmonics up to 12,000 Hz. On the other hand there are countless possibilities to place the microphones. A lot of articles and books are available that describe many ways to mike a piano and in this guide only a few basics can be touched upon.

The only common thing with all ways to mike a piano is the type of microphone used. Typically, a small diaphragm condenser microphone is the best choice, at best a pair to get a stereo-image. Low frequencies are positioned to the left and the high frequencies to the right; this is also how the pianist would hear it.

The lid should be open to get the best results. Instead of only using cardioid microphones, experimenting with omnidirectional microphones that generally sound more natural, will also lead to a better outcome. A standard way to mike a piano is to place the microphones approximately 30 to 60 cm above the strings. One is aimed at the low strings, the other at the high strings. The closer the microphones move towards the hammers, the more attack and brightness is added to the sound; but also mechanical noise such as hammer and damper noise increases.

In any way, finding the “sweet spot” and keeping the microphones placed on this spot is most critical, because moving the mic just a few centimeters can change the sound dramatically.



Further Instruments

The most common instruments for home recording have been briefly described. With this short guide it is not possible to explain miking techniques for every single instrument in detail. Small diaphragm condenser microphones are in general a good choice for further instruments. Nevertheless the placement of the microphone differs for most instruments and therefore testing and experimenting leads you to what sounds best to your ears.

Recording a whole band means having individual tracks and microphones for every instrument or voice. To get the best result the single instrument and vocal tracks should be recorded one after the other. With this you can best avoid scenarios such as the drums being too audible on the guitar track. Miking the whole band with only one or two microphones will not result in good sound quality, but can be considered for capturing ideas.

THE RECORDING ENVIRONMENT

When sound waves are travelling through a room, the direct sound is picked up first by a person as well as by a microphone; afterwards the reflections from surfaces, such as walls, the floor and the ceiling are audible. This ambient sound gives an impression of the room size – is it a small or large room or even a concert hall? A sound signal free of reflections or reverberation sounds unnatural, as it does not exist in nature.

Nevertheless it is the goal to record a signal that is free of ambient sound and reflections – no matter if speech, singing or instruments. If the signal was recorded with ambient sound, it cannot be removed afterwards. Reverberation should be added in the post production, giving the opportunity to choose the desired sound.

What does this mean for home recording and podcasting?

The most important rule is: the closer the mic moves to the sound source, the less ambient noise is picked up and the higher the sound level. Additionally unwanted noise, such as the noise created by a computer fan or the playback music coming from the headphones, is reduced as well.

The boost in low frequencies caused by the proximity effect should be considered when placing the mic very close. The mic should always be aimed directly at the sound source as the level becomes lower when the mic moves away.

The microphone should also not be placed too close to a wall. The reflections from this wall are picked up as well and they degrade the sound quality. Be aware that also simple sheets of paper that you might need to read the lyrics can reflect sound.

The recording room should acoustically be as “dead“ as possible. This means avoiding concrete walls and floors as well as windows. A square shaped room with a tiled floor and concrete walls is surely one of the worst environments for recording. A carpet helps reducing the reflections. Book shelves, curtains at the windows or heavy drapes help to “deaden” the room. Acoustic foam is certainly the best but also a very costly solution. A mattress on the wall or a pile of blankets in a corner of the room is a good starting point.

The room should also be as “quiet” as possible. Higher quality condenser microphones pick up even the quietest signal. A computer is very often a disturbing factor in home recording. A silent processor fan and power supply should be self-evident. A passively cooled graphic card is also recommended. A decoupled hard drive minimizes chassis vibrations and reduces the noise as well.

If the noise cannot be avoided, make sure to follow the already mentioned rules to keep the microphone as far away from the noise as possible and not to aim the mic into the direction of the disturbing sound.

MONITORING

Before you can record or mix good sound, you need to be able to hear what you are getting. In audio terms, monitors allow you to listen to the audio while it is being recorded or edited.

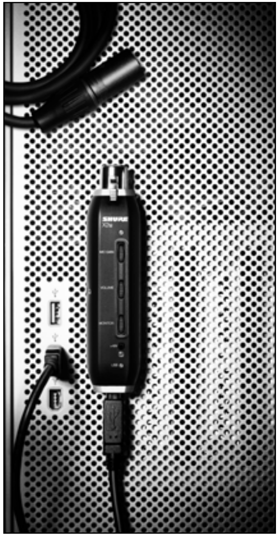
If you're trying to sing or play along with a recorded music track, you need to monitor through headphones, earphones or speakers. With speakers the monitoring signal can also be picked up by the microphone which degrades the sound quality. For that reason they are normally not used for monitoring during recording.

Headphones with a closed back design are most common for this application as the music is not audible and cannot bleed into the microphone.

An alternative are earphones that are inserted directly into the ear and offer the best sound isolation.

If you are trying to sing or play along in sync with previously recorded tracks, you must be able to monitor in real time, with almost no latency. A few milliseconds of latency is not critical, although it can slightly alter your perception of pitch or tone. More than 10 milliseconds of latency can have a noticeable effect on your rhythm and timing. To monitor in real time, you need to tap into the audio signal before it gets converted from analogue to digital and fed into the computer.

A solution for the latency issue is the Shure X2u XLR-to-USB adapter that provides a headphone output, allowing you to monitor without latency. It also offers Monitor Mix Control for blending the microphone signal with the playback audio. The Shure PG27USB and the PG42USB microphones offer the same functionality already integrated into the microphone.



RECORDING DEVICES

Various recording devices for home recording and podcasting are available. A computer with recording software is the most flexible and common solution for home recording.

For mobile recording, battery powered devices are available from analogue one track devices up to multi-track digital products. Recording analogue and a digital post production means transferring the signal into a computer, which can be time consuming. Mobile digital recording devices allow simple editing, fast transferring and sharing of the files through USB or Firewire ports.

The gain adjustment is critical with a digital device. With an analogue signal the distortion is blending slowly in the higher the gain is set and this can lead to a richer sound if used to a low degree. A distorted digital signal immediately creates a disturbing “cracking” sound. As this “cracking” sound cannot be removed in post production, enough headroom should be considered using digital recording devices.

Some digital devices allow users to create both MP3 as well as WAV files. Compressed MP3 files mean less storage space, but also less sound quality. WAV files provide the best possible audio quality, but they require much more storage space and take much longer to upload/download. More information on file formats can be found in the next chapter.

SOFTWARE / DIGITAL POST PRODUCTION

The recording itself is just the first part of home recording and podcasting. The process of editing, mixing and applying effects – also called post production – starts afterwards and is a big part of the project. Below you will find a brief outline of possibilities that recording software and effects processing offer.

Software

There are two ways of post production with the computer: software that is only able to edit one track and multi track software.

With a mere podcast spoken by one person a single track software is sufficient. As soon as you want to add music in the background, multi track software is needed. The same counts for video productions that include voice-overs or background music.

The most common multi track software for PCs is Steinberg Cubase and for Mac it is Emagic Logic. Both are additionally available as affordable, entry level versions. If you don´t want to spend money you can also download free- and shareware, such as Audacity, from the internet.

Multi track recording software is mainly about blending the signals into one stereo mix. Simple mixing and adjustment of levels will in most cases not lead to good sound quality. To get the best result the tracks must be manipulated and changed in certain aspects through effects processing.

Effects Processing

The main impact on the sound is **Equalization** (also called EQ). With an EQ you can emphasize or de-emphasize certain frequency bands, which can either make different tracks stand out from each other or help different sounding tracks sound more similar. E.g. on a bass drum the frequencies above 6 kHz can be cut off and a boost at 80 Hz will create a more powerful sound.

As a general rule, you should try to shape the sound by reducing certain frequencies, rather than boosting others. In particular, excessive boosting of low frequencies is a common cause of less intelligible recordings. These “muddysounding“ recordings often result from the dreaded “smiley-face“ EQ curve, when lows and highs are boosted to the point where the all-important mid-range (critical to intelligibility) is effectively masked.

A **compressor** automatically turns down the signal´s peaks (loud parts) by a preset amount so they don’t cause distortion. Compression also reduces the difference between the loudest and softest note, so the apparent loudness is greater. Example usage: A singer might vary in loudness from very soft to very loud, but the compressor reduces the magnitude of these extreme changes.

As it appears very unnatural to hear sound without reverberation, the **reverb** effect is an extremely important effect. Reverb adds depth to the sound signal and makes it smoother but also more indefinite. It is used in all cases for vocal tracks, but be careful with too much reverb as the signal might diffuse and the vocals lose presence in the whole mix.

Other instruments such as guitars, strings and the piano also sound richer and more natural with some reverb. Impulsive instruments, such as drums, shouldn´t be manipulated with reverb as the attack easily becomes indistinguishable.

There are countless software programs and plug-ins available that add reverb to the signal. The versions that mimic certain environments provide the best outcome but they are also memory and processor intensive and require high powered computers.

Format

After the last step in the project - the mixdown of all tracks to a stereo signal - you need to decide where and in which format to save the files.

When saving your files onto CD, use WAV files with 16 bit and 44.1 kHz as this is the standard CD format. This ensures that you will have a high quality file available if you want to re-use or re-edit the file later for a different project. You should make sure to record all tracks with 44.1 kHz as if you for example record with 48 kHz, the transfer to 44.1 kHz can cause artifacts. A higher bit rate on the other hand is not critical. Especially if the post-production includes many steps, a 24 bit recording makes sense.

Are the files intended to be uploaded on websites or used with an MP3 player, you should consider a compressed format that results in a much smaller file size. The MP3 format with 128 kb has become the standard and is viewed as equal to the original WAV signal. Critical ears though hear a difference and a compression with 196 kb per second results in a better sound quality.

MP3 is not the only format but the most popular. Further formats are e.g. WMA (Windows), AAC (Apple) or Ogg Vorbis.

All formats have one thing in common: the encoding software discards some of the data that is deemed to be unnecessary or redundant and the more data that is discarded, the smaller the file size but the lower the sound quality.

To summarize: it is best to save the initial recordings as WAV files to have the best possible sound quality. While there are programs that convert MP3s to WAV files, they do not make the resulting sound quality any better, because the data that was discarded when the MP3 file was created cannot be “added back in“.

APPENDIX

Microphone Accessories

Pop-Filters

First of all, you need to understand what popping is. When you say the word “pop“ for example, you will hear an explosive breath after the “p“, that is, “po-puh“. Try putting a hand in front of your mouth to even feel the explosive breath. Pops occur most often with “p”, “t”, “d”, and “b” sounds, can be very distracting and even cause distortion on the recording. Most microphones feature built-in pop filters, either as integral foam or metal mesh grille. For professional vocal recordings these built-in filters are rarely sufficient. An external pop filter provides an acoustically transparent shield around the microphone, which breaks up the wall of air before it hits the mic and helps reduce popping sounds.

Shock Mounts

Shock mounts are used to isolate the microphone from vibrations transmitted through the stand or the mounting surface, such as a desktop or floor. A shock mount can reduce or eliminate the “handling“ noise you hear when microphones are moved during a recording session or if the surface upon which the microphone rests is being jarred or vibrated (often called “stand thumps“).

Stands

Stands should be sturdy enough to support the microphone in the intended location and to accommodate the desired range of motion. These accessories come in many shapes and sizes, but the purpose remains the same: to position the microphone in the right place to pick up only the sound you want. Stands also transmit vibrations to the microphone and therefore should be equipped with rubber feet to minimize these vibrations.

Finding the right version for your needs and your microphone is what matters most. For podcasting a small desktop stand might be sufficient. For vocal and instrument pickup a boom arm stand is the best solution as it balances the microphone in mid-air at the exact desired position.

SHURE WIRED MICROPHONES AT A GLANCE

Microphone Overview

PG Series
The entry to Shure.

- The entry to professional Shure quality, reliability and sound
- Analog and digital connectivity options
- Wide range of microphones to suit different applications
- PG27 and PG42 also available as USB microphones
- X2u: USB interface for all XLR microphones

PG48	dynamic	cardioid	tailored
PG58	dynamic	cardioid	tailored
PG57	dynamic	cardioid	tailored
PG81	condenser	cardioid	flat
PG52	dynamic	cardioid	tailored
PG56	dynamic	cardioid	tailored
PG27	condenser	cardioid	flat
PG42	condenser	cardioid	tailored

SM Series
Industry standard. Professional utility.

- Legendary sound for performance and recording applications
- Reduced handling noise and improved gain before feedback
- Rock-solid construction proven through decades of rigorous use
- X2u: USB interface for all XLR microphones

SM48	dynamic	cardioid	tailored
SM58®	dynamic	cardioid	tailored
SM86	condenser	cardioid	tailored
SM87A	condenser	supercardioid	tailored
SM57	dynamic	cardioid	tailored
SM81	condenser	cardioid	flat
SM94	condenser	cardioid	flat
SM137	condenser	cardioid	flat
SM27	condenser	cardioid	flat
SM7B	dynamic	cardioid	flat, tailored

Beta Series
The first choice of professionals.

- Specialized, precision engineered models with high sensitivity
- Maximum isolation and minimum off-axis sound for higher gain-before-feeback
- Unmatched, tour-tested construction and ruggedness

Beta 58A	dynamic	supercardioid	tailored
Beta 87A	condenser	supercardioid	tailored
Beta 87C	condenser	cardioid	tailored
Beta 53	condenser	omnidirectional	flat
Beta 54	condenser	supercardioid	tailored
Beta 57A	dynamic	supercardioid	tailored
Beta 98S	condenser	supercardioid	tailored
Beta 98H/C	condenser	cardioid	tailored
Beta 52A	dynamic	supercardioid	tailored
Beta 56A	dynamic	supercardioid	tailored
Beta 91	condenser	cardioid	tailored
Beta 27	condenser	supercardioid	flat

KSM Series
Premium microphones.

- Wide dynamic range, high sensitivity and low self noise
- Full frequency response and minimized proximity effect
- Shure renown ruggedness combined with premium studio sound

KSM9	condenser	cardioid, supercardioid	tailored
KSM32	condenser	cardioid	flat
KSM44	condenser	cardioid, omnidirectional, bidirectional	flat
KSM137	condenser	cardioid	flat
KSM141	condenser	cardioid, omnidirectional	flat
KSM313	ribbon	bidirectional	tailored
KSM353	ribbon	bidirectional	flat

ribbon · dynamic · condenser · flat · tailored · cardioid · supercardioid · omnidirectional · bidirectional

Application Guide

The following examples show the preferred choices for the listed application (preferred option in bold, alternative options follow).

		PG The entry to Shure. 	SM Legendary microphones. 	Beta The first choice of professionals. 	KSM Premium microphones 
Vocal	Dynamic	**PG58** I PG48	**SM58®** I SM48	**Beta 58A**	---
	Condenser Live	---	**SM86** I SM87A	**Beta 87A** I Beta 87C	**KSM9**
	Condenser Studio	**PG42*** I PG27*	**SM27**	---	**KSM44** I KSM32
	Headset	**PG30**	**WH30** I WH20	**Beta 54** I Beta 53	---
	Choir	**PG81**	**SM81** I SM137	---	**KSM141**
	Ribbon	---	---	---	**KSM353** I KSM313
Acoustic Instruments	Guitar	**PG81** I PG27	**SM81** I SM137	**Beta 57A**	**KSM137** I KSM141
	Brass	**PG57** I PG27	**SM27** I SM58® I SM57	**Beta 98H/C**	**KSM32** I KSM313 I KSM353
	Piano	**PG81**	**SM81** I SM137	**Beta 91** I Beta 27	**KSM141** I KSM137
	Strings	**PG81**	**SM81** I SM137	**Beta 57A**	**KSM137** I KSM141
Electric Instruments	Guitar	**PG57** I PG27	**SM57** I SM7B I SM27	**Beta 57A** I Beta 56A I Beta 27	**KSM32** I KSM313 I KSM353
	Bass	**PG52** I PG57 I PG56	**SM57** I SM27	**Beta 52A** I Beta 56A I Beta 57A I Beta 27	**KSM32** I KSM44
Drums & Percussion	Bass Drum	**PG52**	**SM57**	**Beta 52A** I Beta 91	---
	Snare Drum	**PG57**	**SM57**	**Beta 57A** I Beta 98D/S	---
	Hi-Hat	**PG81**	**SM81** I SM94	**Beta 27** I Beta 98S	**KSM137** I KSM141
	Tom Tom	**PG56** I PG57	**SM57**	**Beta 56A** I Beta 98D/S	---
	Overhead	**PG81** I PG27	**SM81** I SM27	**Beta 27**	**KSM137** I KSM32

* PG27 and PG42 also available as USB microphones

Shure Professional Sound Isolating™ Earphones at a Glance

	SE115 Sound Isolating™ Earphones	SE115m/m+ Sound Isolating™ Headset	SE310 Sound Isolating™ Earphones	SE425 Sound Isolating™ Earphones	SE535 Sound Isolating™ Earphones
					
	The SE115 features 2nd Generation Dynamic MicroDrivers to deliver detailed, warm sound quality with improved bass.	SE115m+ Sound Isolating™ Headset pairs a detailed listening experience with an inline microphone and remote with volume control (m+ exclusively).	High-Definition MicroSpeakers with Tuned BassPort for extended range audio plus enhanced bass.	Dual High-Definition MicroDrivers with dedicated single woofer and tweeter per channel for accurate and balanced sound.	Triple High-Definition MicroDrivers with dual woofers and single tweeter per channel for spacious sound with rich bass.
Sensitivity	105 dB SPL/mW	105 dB SPL/mW	111 dB SPL/mW	109 dB SPL/mW	119 dB SPL/mW
Impedance	16 Ω	16 Ω	28 Ω	22 Ω	36 Ω
Frequency Response	22 Hz - 17,5 kHz	22 Hz - 17,5 kHz	22 Hz - 19 kHz	20 Hz - 19 kHz	18 Hz - 19 kHz
Weight	30 g	30 g	28 g	32 g	33 g
Cable Length	45/ 136 cm*	149 cm	45/ 136 cm*	157 cm	157 cm
Color	Blue/ Red/ Pink/ Black	Black	Black/ White	Clear/ Metallic silver	Clear/ Metallic bronze

* Extention Cable

Shure Professional Headphones at a Glance

SRH240
Professional Quality Headphones

The SRH240 is designed with a wide frequency range for full bass and clear highs, ideal for general listening.

SRH440
Professional Studio Headphones

The SRH440 for home or studio recording. They reproduce accurate audio across an extended range and provide exceptional sound reproduction and comfort.

SRH840
Reference Studio Headphones

Engineered for critical listening and recording. It delivers accurate sound reproduction in demanding studio applications

SRH750DJ
Professional DJ Headphones

The SRH750DJ features 50 mm neodymium dynamic drivers tuned to deliver high-output bass with extended highs. 90° swivel ear cups for easy one-ear placement.

	SRH240	SRH440	SRH840	SRH750DJ
Design	Closed-back, circumaural	Closed-back, circumaural	Closed-back, circumaural	Closed-back, circumaural
Transducer Type	Dynamic	Dynamic	Dynamic	Dynamic
Sensitivity	107 dB SPL/mW	105 dB SPL/mW	102 dB SPL/mW	106 dB SPL/mW
Impedance	38 Ω	44 Ω	44 Ω	32 Ω
Input Power	500 mW	500 mW	1000 mW	3000 mW
Frequency Response	20 Hz - 20 kHz	10 Hz - 22 kHz	5 Hz - 25 kHz	5 Hz - 30 kHz
Weight	181 g	272 g	318 g	227 g
Cable Lenght	2 m	3 m	3 m	3 m

Glossary

Balanced/ unbalanced circuit – An unbalanced output carries the signal on a single conductor (plus shield). Influences on the cable (like the humming of a parallel power cable) are audible. When using a balanced output the signal is carried on two conductors (plus shield). The signal on each conductor is the same level but the opposite polarity. A balanced microphone input amplifies only the difference between the two signals and rejects any part of the signal which is the same on each conductor.

Cardioid microphone – See page 18

Compression formats

AAC – Apple compression format

FLAC – Lossless compression format

Ogg – Vorbis compression format

MP3 – Most popular compression format.

WAV – Digital audio format without compression and discards

WMA –Windows compression format

Compressor – A device or software feature that controls varying signal levels by reducing the level of loud sounds.

Condenser microphone – See page 15

Decibel [dB] – A number used to express relative output sensitivity. It is a logarithmic ratio.

Dynamic range [dB] – The range between self noise and the maximum sound pressure level. The wider the dynamic range the more accurate differing sound levels can be reproduced.

Dynamic microphone – See page 15

Electret (permanently biased) condenser microphone – The microphone capsule (membrane and backplate) of a condenser microphone requires polarizing voltage to charge the condenser element. An electret (a synthetic polarized material) attached to the backplate, the polarizing voltage does not need to be supplied externally. Nevertheless, an electret condenser microphone also requires power (by battery or phantom power) to operate the preamplifier.

EQ/Equalization – Equalization or tone control is used to shape frequency response (and sound quality) in some desired way.

Feedback – During the normal operation of any sound system, sound produced by the loudspeakers can be picked up by the microphones, re-enter the system and become amplified. At certain points this can cause the system to create a noisy, sustained "howl" known as feedback.

Frequency – The rate of repetition of a cyclic phenomenon such as a sound wave. Usually measured in Hertz (Hz).

Impedance [Ω] – In an electrical circuit, opposition to the flow of alternating current, measured in ohms. A high impedance microphone has an impedance of 10,000 ohms or more. A low impedance microphone has an impedance of 50 to 600 ohms.

Omnidirectional microphone – See page 18

Phantom Power – A method of providing power to the electronics of a condenser microphone through the microphone cable.

Proximity effect – See page 19

Self noise [dB] – The self-noise or equivalent noise level is the sound level that creates the same output voltage as the microphone does in the absence of sound. This is the lowest point of the microphone's dynamic range, and is particularly important with recording sounds that are quiet.

Sensitivity [mV/Pa] or [dBV/Pa] - The electrical output that a microphone produces for a given sound pressure level. In most cases sensitivity is measured with a sound pressure level of 94 dB (1 Pascal). The higher the sensitivity, the “louder” the microphone.

Small and large diaphragm – The terms small and large diaphragm are used with condenser microphones. A large diaphragm has a diameter of at least 1 inch (2.54 cm). Large diaphragm microphones are popular for vocal recordings as they add harmonics to the sound which makes voices sound smoother. Small diaphragm microphones feature a flat frequency response and sound more natural. This is why they are popular for instrument recordings.

Stereo – Two channels of audio, left and right, which can be used to simulate realistic listening environments.

Supercardioid microphone – See page 19

THD – total harmonic distortion [%] – The total harmonic distortion, or THD, of a signal is a measurement of the harmonic distortion present and is defined as the ratio of the sum of the powers of all harmonic components to the power of the fundamental.
Transducer type – See page 15

USB – An acronym for Universal Serial Bus, a standard designed to allow many different types of devices to connect to a computer using a standardized interface. USB also can provide power to low-consumption devices, negating the need for external power supplies. There are currently two standards: USB 1.1 and USB 2.0. For audio applications, USB 2.0 (which offers much faster data transfer rates) allows many more channels of audio to be streamed to the computer at once.

Want to know more?
Additional literature can be found under the category “Tech Support” on www.shure.com/proaudio

IMPRINT

Contact

Shure Europe GmbH
Headquarters Europe,
Middle East, Africa
Wannenäckerstr. 28
74078 Heilbronn, Germany

Phone: 49-7131-7214-0
Fax: 49-7131-7214-14
Email: info@shure.de

Technical Support:
Tel.: 49-7131-72 14-30
Email : support@shure.de

SHURE AGAINST COUNTERFEITING

Did you know many popular Shure models including the SM58® and Beta 58A are illegally manufactured and sold around the world as authentic Shure products?

Despite all superficial similarities to authentic Shure products, counterfeits, on average, use much lower quality materials and are very unreliable, much less rugged and offer significantly lower performance and sound quality. Counterfeits are also not covered by Shure’s warranty policy should you need it.

While Shure is taking action to protect you and our brand, there are things you can actively do to reduce the chances that you purchase a counterfeit:

- Be a wise shopper. Familiarize yourself with signs of counterfeit products, be cautious of incredibly low prices offered by on-line auctions and merchants and, when possible, inspect merchandize before you buy.

- Buy only from authorized Shure dealers. You can find a list of authorized dealers and distribution centers on the Shure websites.

Shure Europe GmbH
Headquarters Europe,
Middle East, Africa
Wannenäckerstr. 28
74078 Heilbronn, Germany

Phone: 49-7131-7214-0
Fax: 49-7131-7214-14
Email: info@shure.de

www.shure.com

 AL1663SE Printed in Germany 3/ 2010