All about harmonica microphones…

And then some

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Introduction

If you're a harmonica player, sooner or later you're going to want to play through a microphone, to be better heard, to change your tone, to record.... or some combination thereof. I'm a semi-pro harp player, and through my business, BlowsMeAway Productions, I have had the chance to play through, repair, modify and build a wide range of microphones. Over the years, I've learned a few things. These are my thoughts about what you should know in order to choose the mic (and other components of an amplified system) that's right for you, and how to get good tone through it. I'll define terms as they come up - I think this is better than presenting a glossary of boring terms up front. But let's just agree on the obvious - a "microphone" is a device that converts acoustic energy (sound waves, changes in air pressure) to electrical energy, which in turn allows for electronic amplification. This conversion is the job of the microphone's "element". In practice, when we talk about mics we tend to care not only about the element, but also the physical housing, or "shell" for the element.

The most common housing styles are commonly referred to as "Stick mics" or "Bullets". Stick mics tend to have smaller heads and long handles. The most popular among stick mics today can be found on just about any stage in the world - the Shure SM57 and SM58. Bullet mics are larger housings, shorter and bigger around. Many taper to a point in the rear so they have a "bullet" shape. So - when we talk about mics, we're talking about both the housing and the element, and both are important.

Acoustic vs. Amplified

In choosing a microphone, the most important question of all is “what do you want to sound like?” There are two classes of sound, with infinite variety in each class. If you’re a classical musician, you’ll probably stand in front of a stand-mounted microphone and you want the amplified sound to cleanly reflect the tone of your playing – giving the closest representation of what it would be like to be in the same room with you, with no microphone or amplifier at all. We call this kind of sound “acoustic” even though it may in fact be picked up with a microphone and amplified.

When I talk about acoustic playing, I do not mean “not amplified”. I do mean, however, that the microphone is in *free air* – that is, on a rack or a stand, and *not* in your hands.

If you’re a blues musician, you might want the fatter, more distorted sound that is so prevalent in the genre. We call this style “amplified” style as opposed to “acoustic” tone. (Hey, I didn’t write the rules – sorry if it is confusing!) In this case the microphone is almost always hand-held. Jazz, Country, Bluegrass and other styles often find themselves between these two extremes.

Acoustic tone

Acoustic amplification is straightforward and can be effectively done with modern, relatively inexpensive gear. A standard vocal mic like Shure's venerable SM58, plugged straight into the PA, is a safe bet and usually sufficient. Lapel mics, which are usually of the "electret condenser" type, can also be used, although most harp players prefer a dynamic or ribbon mic for acoustic playing. The PA (short for "Public Address") is a very clean-sounding amplifier designed to accommodate multiple microphones, each with their own volume and tone controls. Most PA's have simple effects like reverb that can be added to each microphone channel, and deliver sound to the "house" speakers (for the audience to hear) and to "monitor" speakers which are on stage (to help musicians hear themselves and each other).

Good acoustic tone is largely a function of the player's tone in the first place. So for now I'm not going to talk about acoustic microphones specifically. I will talk about them when they come up in conversation. However one essential piece of wisdom is that your distance from the microphone makes a huge difference in volume. It is, unfortunately, rather easy to hurt people's ears when you walk up to an acoustic mic set up for someone to sing through at a distance of 6 inches, and you play loud harp right up against it, or worse, pick it up and cup it. The secret to good acoustic tone through a microphone is to learn to play quietly! If you sing and play through the same mic, a volume control on the microphone is a very useful tool.

Where does "amplified tone" come from?

To understand the amplified sound, we need to talk about the interaction between the player, the harmonica, the amp and the mic. Where does that big fat gritty sound come from?

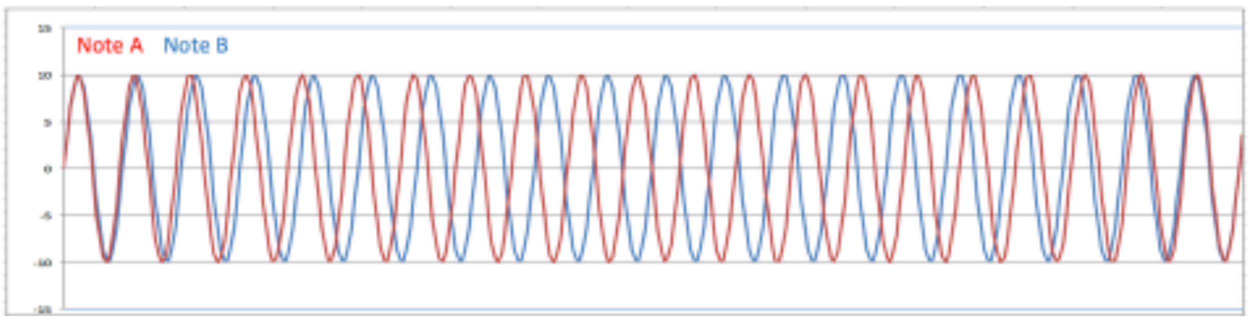
Your harmonica makes bass! "Difference Tones" explained.

Many people don't realize that when you play two notes, a 3rd note called a "difference tone" is produced. That note is far lower in pitch than either of the two fundamental notes. Without amplification you probably don't hear it at all. But with a good mic and amplifier? You absolutely do. How does this happen? Bear with me, because I think this is really cool. It's physics! Don't let that scare you.

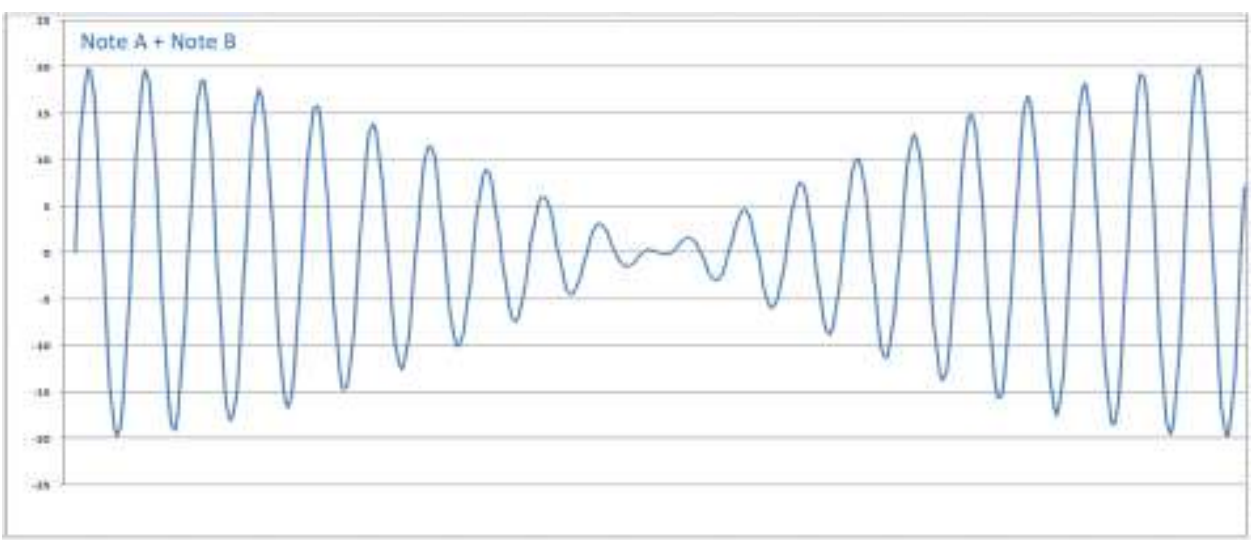
Difference tones are just faster beat notes

Many musicians are familiar with the phenomenon of “beat notes”. When we hear two notes that are *almost* the same in pitch, we also hear a sort of throbbing sound – which is actually the total sound pressure slowly increasing and decreasing while we’re listening to the two fundamental notes. What’s happening?

Each note produces a wave of pressure that alternates between positive air pressure (speaker cone moving out, ear drum moving in) and negative pressure. When the two notes are almost in tune, they go in and out of phase with each other fairly slowly. The following chart demonstrates this. In this example, Note B is just slightly lower in pitch than Note A.



You can see that at the left and right ends of this chart, the waves are almost in phase. But the blue note (or pressure wave) is slightly longer. At the center of the chart, when the red wave is making maximum positive pressure, the blue wave is making maximum negative pressure. So what do we hear? We hear the sum of the pressure carried by these two waves. When the two are in phase, the sum (“amplitude”, or “volume”) is twice as big. When they’re out of phase, it is 0.



Now, Note A and Note B are vibrating at a few hundred times a second. But you can see they're going in and out of phase with each other much more slowly if they're very close in pitch. We hear this change in overall amplitude as "beat notes." Most of the time we hear beat notes the waves are going in and out of phase just several times a second – something less than 20 cycles per second.

When Note A and Note B grow further apart in pitch, the number of times per second that the sound waves go in and out of phase with each other increases. Said another way, the speed of the "beats" increases. Now it might be happening at 40 cycles per second (or "Hz".) We stop hearing "beat notes" and start hearing a new note!

And that (drum roll please) is where the bass comes from!

So now we know that we get new bass notes when we play, *but only if we play more than one note at once*. So let's get back to amplification and harmonics and all that good stuff.

What are "Harmonics?"

Now amplified sound is more than just adding those bassy difference tones into the mix. It also comes from a combination of the microphone and the amplifier's tendency to add distortion in the form of even more "manufactured" notes called "harmonics." Unlike difference tones, which are lower in pitch than the notes you're playing, harmonics are "overtones" that are higher in frequency than the fundamental signal.

Harmonic distortion can occur simply because the amplifier itself is operating beyond its "clean" volume range, but as players we have control over the degree to which it occurs by our own technique and microphone choice. We can overdrive the microphone, through our own playing volume and more importantly a technique called "cupping", so that it sends a distorted signal to the amplifier. The amplifier responds by amplifying that distortion and more readily adding its own. Of course, amplifiers vary a great deal in the degree to which they add harmonics, and in the quality of that sound.

Amplifiers - Tubes vs. Transistors?

An important characteristic of tube amplifiers is that they produce "even ordered harmonics" and "analog clipping" when they distort. These are felt to be pleasing to the ear. Solid state amplifiers tend to produce "odd ordered" harmonics and "square wave" clipping when they distort. These can sound very harsh. As a rule, either kind of amp played cleanly/acoustically will sound fine. But overdriving a solid state amp, such as any modern PA, or some modern guitar amplifiers, usually results in harsh sound. Another very important advantage of the tube amplifier is our ability to manage feedback through tube substitution to lower the amp's gain. More on that later.

Cupping:

Good amplified tone starts with the player's tone, and is accentuated by microphone technique. Cupping is an art; a learned skill that is neither obvious nor easy in practice. Properly done at its extreme, no air you suck or blow can escape "the seal". If you can do this perfectly, no sound can come out of the harp at all! So we always allow a slight leak, but until you can cup so tightly that you literally can't get a sound from the harp, you haven't perfected this technique. In reality it is very hard to do, and it is hard to even come close at first. Note that to accomplish such a seal, you must not only create an airtight seal between the rear of your harp and the microphone, you must also seal off the open holes on the *front* of the harp - otherwise the sound pressure is free to travel under the cover plates, through all the open reeds and out/in the other open holes on the harp. **Leakage out of the front of the harp is the single most under-appreciated cause for thin tone.** You can learn the technique acoustically before ever attempting it amplified. Sealing the front of the harp will increase the depth of your "wah".

When the seal with a microphone is very good, the air pressure changes are effectively "coupled" to the microphone's diaphragm in a way that is very, very different from the normal "free air" mode in which mics were designed to operate. The result is a very strong, distorted signal sent to the amp, that has more low frequency components and fewer high frequency ones.

To choose a good harp mic, you must understand what cupping is - as different microphones will respond in dramatically different ways to the technique.

Why are some mics more responsive to cupping?

Factor 1: Mic shape and size:

When a mic's physical shape or size makes it hard to achieve a good seal, the effect of cupping will be less.

Factor 2: Headroom

When a mic has a lot of "headroom", which is the amount of sound pressure it can tolerate before it distorts, it will obviously distort less. A perfect example of this is the Audix Fireball, which has something like 140dB of headroom. A brilliant feat of engineering, but it makes almost no difference to your tone whether it is cupped or not.

Microphone shapes

There are literally hundreds, if not thousands of different kinds of microphones to choose from. Mics vary in cost, size, purpose, tone, and shape - all of which affect us. For amplified tone, one of the most important variables is the mic's shape. There is little about the shape that affects the mic's tone in free air. However everything about its shape affects your ability to cup it easily and effectively. People's hands and muscles differ, so one size most definitely does **not** fit all.

Some mics have holes all over the place and are virtually impossible to cup effectively. A stick mic with a ball end like the SM58 (lower mic in photo) can also be difficult for smaller hands. The smallest diameter stick mics, like Shure's SM57 (upper mic in photo), are easy for some to cup, yet cause hand cramps for others.

And their length gives the weight of the cable leverage to pull down on the end of the mic – another fatigue factor. (The weight and length of Shure stick mics was one of problems I set out to solve when I developed the Ultimate Series Microphones.) The following photograph illustrates this:

The Ultimate series mics shorten and lighten popular stick mics, and add a volume control – making them much nicer harmonica microphones than the generic equivalents from which they are derived.

Bullet mics are also very popular among harp players.

Photo courtesy of kingtonemics

Bullet mics are named for their shape. These were popular and inexpensive mics beginning in the late 1940's, which resulted in harmonica players picking them up. Players soon discovered the shape lent itself to being cupped and therefore delivering big fat tone.

The largest diameter bullet in popular use is the current model of Shure's Green Bullet, called the 520DX. It is hard for people to cup well unless they have fairly large hands. It is also one of the heaviest mics available - and when you perform a three or four hour show, hand fatigue is a real factor worthy of your consideration. Additionally, some mics like the popular JT30 or Hohner Blues Blaster have bumps around the circumference that make tight cupping very uncomfortable. Among vintage bullet mics, there are many shells that are a little smaller. The Electro-Voice 630 and M23/43, the Astatic JT30 and T3, and the Shure 707 are all excellent shell choices and are popular among "do it yourselfers" as mic project starting points.

And of course BlowsMeAway's wood mics are among the most comfortable to play thanks to their reasonable diameter and light weight.

Charlie Musselwhite's BlowsMeAway Wood Microphone

Introduced in 2015, the new Bulletini™ microphones are even smaller in diameter. They contain an element specially designed for harp – so the combination of a mic that's easy to cup and the tone of an element that is designed to help roll off the highs and break up easily – makes this a very popular microphone.

The BlowsMeAway Productions Bulletini Microphone – compared to Shure Green Bullet in diameter

Another variable is where the mic's shell places the element relative your harp. Some shapes will hold the element further forward, others further back. Further forward reduces the size of the air cavity between harp and element, reducing the tonal changes from cupping. This effect is slight but real.

And yet *another* variable is the amount of space *behind* the element. This only matters for dynamic elements – crystal elements are completely closed on the back side. But dynamic elements' bass response is controlled to some extent through the volume of air space to the rear of the element.

Due to this interaction with the space behind the element, dynamic elements are also much more prone to feedback if the space to the rear of the element has a way for sound from the outside to enter easily. A common example of this is when someone removes a built-in volume control and leaves the mounting hole for the potentiometer exposed. Another is with mic that have a swivel-mounted base, like the Astatic T3. The original crystal element didn't care about the big hole where the swivel mechanism entered the shell, but dynamic elements do. That's why T3 mics which have been modified to house a Shure CM or CR tend to feedback more than normal.

So you see - a mic's shape makes a real difference! You'll have to experiment to find what works best for you. The other important variable, of course, is the element.

Microphone Elements

There are many kinds of elements in all of microphone-dom, including ribbon mics, condensers, electret, crystal and dynamic. Acoustic players may well use any of these. However, with the exception of the recording studio, amplified players will only be concerned with dynamic and crystal elements.

There is no such thing as a microphone element that was designed for use by harmonica players*. Every microphone uses an element that was designed for more general purposes. As a rule, the more expensive a mic was when new, relative to other mics of its vintage, the better it performs as a general purpose mic. To engineers, this means it has better frequency response (able to "hear" higher and lower sounds), flatter frequency response (no particular frequencies are made significantly louder or softer), and/or more "headroom" (the ability to tolerate higher sound pressure levels without distorting).

In this case "better" indeed usually **is** better for acoustic players. Beyond that, choosing an acoustic microphone requires a very critical ear and a lot of listening to hear any difference between mics. Among modern instrument and vocal mics, the differences lie in sensitivity, and subtle changes in frequency response. Some mics tend to accentuate certain frequencies that can work for us or against us.

But here's a shocker: for amplified, bluesy ballsy tone, **better is worse.** Too much high frequency response can sound harsh, even hurt people's ears. The reason vintage bullets are so popular is that their frequency response typically falls off at as low as 5000 Hz – even though some humans can hear as high as 20,000 Hz.

* **Well, that was true until 2015**, when BlowsMeAway developed "The Heumann Element™". The Heumann Element is a new high impedance dynamic element developed specifically for harp players. It is in fact the ONLY element ever developed specifically for harp. It is made from a modern vocal mic element to which both mechanical and electrical modifications are made. It has big, fat tone with plenty of bass. The Heumann Element is the element in the Bulletini microphone and is available for use in other shells as well.

Headroom

High headroom also works against us – it prevents us from overdriving the microphone. The Audix Fireball, marketed as a good harp mic, has amazingly high headroom - 140dB according the spec. And it **is** a good harp mic for playing acoustically, or wanting to avoid a distorted input signal for other reasons, such as playing through an amp-modeling pedal. But the difference in tone between cupped and uncupped with this mic is **far less** than it is most other microphones. In the case of amplified tone, older/cheaper is often better. The venerable original Shure Green Bullet and the original Astatic JT30 are prized by harp players – because they give us that "old school" sound. They are, in many cases, what our harmonica heroes used. Do you know why they chose them in the first place? Because they were **cheap**, even when they were new! The JT30 was the entry level microphone in Astatic's line, and came on the market at a price around $6.

Which are the most desirable Dynamic elements?

The most desirable dynamic elements among amplified players are the vintage Shure "Controlled Magnetic" and "Controlled Reluctance" elements ("CM" and "CR".) Yes, CR's and CM's are dynamic elements – but because they are special ones we usually refer to them by their marketing names instead of just lumping them in with all the other "plain old" dynamic elements. The design, and tone of these elements changed over time.

Chronologically, it went like this, from mid-forties to today:

Late 40's------->Early 50's-------> Late 50's------------->Early 70's----------->Mid 80's

Black label CR --> White label CR --> Single impedance CM -->Dual Impedance CM -->520DX Dynamic

The Single impedance CM was in the original Model 520 Green Bullet. Earlier Shure bullet models including the 440, 707 and 520SL came with CR elements and later CM. These all had a thin metal foil diaphragm. The dual impedance one was in the 520D, and a modern, mylar-diaphragm dynamic element (not a CM at all) is in the current 520DX model produced to this day. With each successive generation, the elements got a little cleaner, gained a little more high frequency response, and a little more headroom (ability to withstand high sound pressure levels without distortion.) Of course since what we crave is the "right" kind of distortion, we often prefer the older elements. The Black Label CR's are considered the "Holy Grail" of these elements and are preferred by many pros.

Much has been written about Shure's Bullet elements; Dave Kott's excellent site http://www.greenbulletmics.com/ has more information than you'll ever need so I won't repeat it here. Suffice it to say that these are great elements, and because of their "magnet and coil" construction, they have tended to last much better than crystals. Many players are still using the dynamic elements from vintage Electro-Voice and Turner microphones as well, although in my opinion, they don't sound as good as the Shure CM and CR's.

Dynamic elements are the most common among modern microphones as well like the Shure SM57 and SM58. There is a huge range of dynamic elements. The newer they are, the more likely they are to appeal to acoustic players and the less they will to amplified players. This is because they have extended high frequency response that can be harsh and they don't tend to "break up" as easily, making it harder to induce distortion at the amp. However there are exceptions. The Shure SM57, for example, actually breaks up very nicely when cupped, and makes an excellent harp mic.

And the new The Heumann Element has gained a large following from beginner to pro. Please don't take my word for this. You can hear what dozens of customers think at http://www.blowsmeaway.com/bulletini_quotes.html

Which are the most desirable Crystal elements?

The crystal elements most craved by harp players were those first made by Brush, Turner, Astatic and Shure. Shure and Astatic both manufactured their elements under license from The

Brush Development Company, who had several patents on the technology at the time. Unfortunately those elements are all but extinct. The Astatic MC-151 element was made for a much longer period of time, but has been out of production for many years. They are available (I still collect them for my customers) but their prices are going through the roof. Over time, any crystal absorbs moisture and softens until it literally falls apart. (If you buy an "untested" crystal on eBay, it is practically guaranteed not to work. Often, if you shake them they rattle. This is what is left of the crystal bouncing around inside the element.) Along the way, they become more and more susceptible to damage from drops, temperature extremes, or even very aggressive playing. (I once ruined a crystal, which undoubtedly was getting ready to go anyway, by drawing really hard with a tight cup. The vacuum pulled too hard on the diaphragm, and broke the connection to the crystal.)

Crystal elements are still manufactured, but with the goal of extremely low cost. They differ significantly in design, with much smaller diaphragms, and in tone. The current Hohner Blues Blasters contain modern crystal elements, as does the Hohner Roadhouse JT30. These are good value-for-money and can produce decent tone in the hands of a good player. But they simply don't have that characteristic "crystal honk" that the old ones had.

Dynamic and crystal: what's the practical difference?

Both crystals (and their close cousins, "ceramics") and CRs/CMs have a great tone for amplified blues-style playing. It is very hard to find words to describe the difference, but to me a good crystal has a slightly nasal "honk" to it that the CR or CM doesn't have. CR/CM elements have more bottom end and can sound a little fatter/richer. The only way to know is to try good examples of each.

However there are some other important factors to consider. Crystals have a lot going against them. The good ones were made 70 years ago and are at the end of their useful life. Many have died, most are dying, and a single drop can kill a good one. They are also extremely high-impedance elements. This means connecting the to a low impedance input will dramatically reduce their tone. Just as turning on all the electric stuff in your car makes the engine work a little harder to keep turning the alternator, lowering the resistance across the mics' terminals is like dragging your foot on the brake. Not only does the total output drop, but the frequency response changes too. For this reason some custom harp amps are made with a 5 megohm input specifically for crystals. An amp with 50K input impedance will suck the tone right out of the best crystal. Anything you add between the mic and the amp, such as a volume control or a long cable, can lower the impedance as well. A good amp and good volume control will still let you get good tone, but you have to be aware that this is an issue and manage it. Finally, if your amp hums when you connect a cable that has no mic at the end, you can expect hum when you connect a crystal-element mic. A dynamic element may well reduce or even mute the hum.

Dynamic elements (including vintage ones) are practically bulletproof. They have lasted well all these years, so they are less expensive and more plentiful. Their impedance is lower so they stand up well to volume controls, pedals, splitters, etc. (Ultimately, though, if the input impedance drops too far you will suck tone from *any* element.) And you can drop them (protected, of course, by a proper gasket inside a microphone shell) and they don't break!

Side note: How do they actually work?

A dynamic element uses the electromagnetic principle to convert sound into electrical energy. A diaphragm is an extremely thin membrane that vibrates in response to sound waves. The diaphragm in a dynamic mic either directly moves a coil around a fixed magnet, or moves a pin that in turn moves inside a coil. Either way, the movement generates an alternating electrical current. A dynamic microphone and a speaker are almost identical in principle. With the speaker, you take electric energy and convert it into sound by sending the current through a coil, which then wants to move relative to a fixed magnet. The speaker cone is attached to the coil. Voila - sound! A microphone is the reverse - you take sound energy and turn it into electrical energy by moving a coil past a magnet. A speaker can actually be used as a microphone, and a dynamic microphone can be used as a speaker (think headphone speaker.) Please **don't** experiment with your good harp mic elements to see how well they work as speakers. They don't work well, and the chance of your damaging them is very large.

A crystal element relies on an entirely different principle called the "piezoelectric" principle - the diaphragm is connected to a crystal of "rochelle salt" or a manmade ceramic with piezoelectric properties. Mechanically deflecting the crystal generates an electric current. The lighter on your gas barbecue uses a crystal too. You press down on a button that mechanically bends and "snaps" a crystal (without breaking it) - and with no battery required, a spark is generated. Crystal elements are rarely made these days, although they can be made incredibly cheaply so they still exist for entry level applications and are found in some production microphones. The new ones, unfortunately, don't have the tone of the older ones - and the older ones are becoming very expensive. If you're an engineer or a scientist and want to read some really deep theory (and interesting history, engineer or not) on the entire subject – check out http://mysite.du.edu/~jcalvert/tech/microph.htm

How do I get one?

Ah, the million dollar question. If you like to gamble, you buy them on eBay. If you don't, buy them only from a reputable dealer who knows harp and knows elements. eBay is filled with deception and ignorance. If you see a crystal element available (usually in a mic shell) with a description that says "I have no way to test it" you should assume it is dead. Of course this isn't always the case, but it is generally true. Crystals don't always fail all the way at once. Their output can simply get lower and lower. Or they will respond unequally in response to positive vs. negative sound pressure (blow vs. draw.) So even one that "works" may not work well at all. The only way to know is to try an element by hooking it to an amp. And the only way to know then whether it is in truly great shape is to know how loud they're supposed to be. I buy crystal elements to provide to my customers. I charge a high price for them. Why? Because I've had to eat the cost of a lot of lousy elements I will not resell. If you buy from me, you get a good element, or your money back!

Does the "ohm reading" matter?

You'll often see eBay vendors showing you the resistance measured across an element. Rule number 1: never hook an ohmmeter to a crystal - you can damage the crystal this way. But across a dynamic element, the fact that the resistance is neither 0 ohms (a short) or infinite (open) tells you that the coil in the element is still working. The value will also tell you something about the element's impedance. 50-100 ohms indicates a low impedance element, somewhere around 1000 ohms indicates a high impedance element, and somewhere in between can indicate a medium impedance element. These are rare, but should be avoided, as it is difficult to find a proper impedance matching transformer for them. More on impedance-matching transformers later.

The ohmmeter's reading, however, will *not* tell you much about the tone of a particular element. Do not believe that 1.1K ohms is better than 1.2K ohms, or vice versa – it simply isn't true. This isn't to say that there isn't a physical meaning to these numbers, only that other factors are much more important when resistance readings are within a few hundred ohms of each other. A valid ohms reading is also not proof that the element is good. An element that's corroded or squashed may not work or will work poorly, even if the coil is OK.

What's Impedance?

The microphones we are talking about are referred to as "high impedance" or "low impedance." In general, a vintage bullet mic is a high impedance device and a modern vocal mic is a low impedance one. This is not always the case. You need to know what you have and how to use it.

By definition, impedance is "resistance to an AC signal" – AC means alternating current – in this case the signal from your mic, which is an electrical picture of the sound going into it. Power in electrical terms is all that matters, but you can have the same amount of power with low voltage and high current, or high voltage and low current. But you don't need to know that. All you need to know is this:

1) Microphones and amplifiers work best with certain load or drive levels, respectively. For this reason, it is desirable to "match" the impedance of the microphone to the impedance of the device it is connected to – an amp, a foot pedal, a wireless transmitter …
2) Impedance matching transformers are simple devices that can "match" a low impedance mic to a high impedance load, or vice versa. Due to the connector sexes involved, devices that connect low impedance mics to high impedance inputs are usually referred to as Impedance Matching Transformers, while devices that connect high impedance mics to low impedance inputs are referred to as "DI (for "Direct Input") boxes". But make no mistake – both are in fact impedance matching transformers!

Figure 1A typical impedance matching transformer

3) Low impedance systems came later than high impedance ones, and were developed to enable much longer cable runs and better reject noise (like hum) that gets injected via the cable.
4) There is no inherent difference in tone or feedback rejection between low and high.
5) Low impedance systems are almost always wired with "XLR" jacks and plugs. This is a worldwide industry standard.
6) Many different kinds of connectors are used for high impedance devices, including XLR. However when XLR is used, the cable-to-pin wiring is not the same as it is for low impedance XLR. **Mixing low- and high-impedance cables and mics just because you can physically connect them together can lead to poor performance.** When you see a mono 1/4" jack or plug (like guitar players use) it's a safe bet that you're looking at a high impedance device. The same is true for the "screw-on" connector that was made by Switchcraft and Amphenol, was found on many vintage mics and is still produced today.

Low Impedance Low impedance High impedance High Impedance




7) There's nothing wrong with connecting a high impedance mic with an XLR connector to a high impedance load (like an amp) with a cable that has XLR at one end and a 1/4" connector at the other. There **is** something wrong, however, with using that same cable to connect a low impedance mic to the amp. First, there is an impedance mismatch and neither the amp nor the mic will perform as well as they could. More importantly, high impedance XLR and low impedance XLR cables are wired differently (called "unbalanced mode" and "balanced mode", respectively.) When you use an XLR→1/4" cable with a low impedance mic, you will get **less than half of the mic's output**! The proper way to connect a low impedance mic to a high impedance amp is to use a low impedance cable (XLR to XLR) together with an impedance matching transformer.

Feedback - are some mics better than others?

Feedback is that awful loud screeching, humming or whistling sound a system makes when a microphone picks up the sound from the amplifier's speaker and sends it back to the amplifier for further amplification. Every system (i.e. microphone plus amplifier) has a feedback threshold. Turn the volume up loud enough and feedback occurs. Keep the volume below that point and it doesn't. The most common cause of feedback is needing more volume than your system can produce without encountering it! An amp that sounds extremely loud in your living room may be completely inaudible on stage. So you turn it up. And it feeds back. It is a law of nature. You can't bring a knife to a gun fight – whatever your mic/amp combo are – they WILL feedback if turned up too far!

Unfortunately we often need to have our volume very close to the feedback threshold in order to be loud enough, and feedback can come and go as conditions change. But some systems *are* less prone to feedback than others. And some microphones are less prone than others. Factors influencing feedback are:

1) Volume required exceeds pre-feedback power available

2) Gain of the amplifier
3) Sensitivity of the microphone
4) Distance of microphone from speaker (amplifier or PA, house or monitors)
5) Tone settings
6) Player technique
7) Directionality of microphone

What can be done about feedback?

Believe it or not, microphone choice is not as important as you might think. Many players believe one microphone feeds back more than another, only because that microphone is louder/hotter/more sensitive than the other. The only "fair" way to judge one mic vs. another is to adjust the amplifier's volume so that the sound out of the speakers is equally loud from mic A to mic B. If mic B is louder to begin with, it will appear to feed back sooner if this isn't done. With all that said, extremely sensitive mics will indeed be more feedback prone, but I have found these to be fairly rare among mics I've seen. Every once in a while I encounter a bullet mic that just seems to feed back very quickly compared to others of the same brand and model. Sometimes this can be attributed to holes in the mic, but often the only solution is … don't use that mic!

Vocal mics and instrument mics are designed to be "directional" – to pick up more sound from the front than from the sides or the rear, specifically to help avoid feedback on stage. So a well-designed microphone can help with a feedback problem – **but only when it is in free air – in a rack or on a stand, and *not* in your hands.** Once you pick up a microphone and cup it, all bets are off. If you hand hold a mic, feedback will be more of an issue because the mic is moving around and facing different directions. When you move your "cup" away from your face, your other hand can act like a satellite dish and can catch and reflect more sound to the mic than if it was in truly free air. Many players learn to manage this by pressing the face of the mic against their chest while not playing. Ideally, you should be set up just a little further from "the hairy edge" of feedback so it isn't too difficult to manage.

When you play acoustically, you can help to control feedback by ensuring that little to none of your sound is coming through your monitors. Depending on the frequency of the feedback, equalization (tone control) can help as well.

When you're playing amplified, all of the above applies. A microphone is designed specifically to pick up sound waves from the air, so it is a much more sensitive and troublesome feedback device than a guitar string. Consequently guitar amplifiers (and most harp amps start out as guitar amps) are set up with much more gain than we need. This allows the guitar player to get Hendrix-like feedback when he or she wants it, but it can be a nightmare for a harp player. More on gain in a moment, but first let's talk about Issue #1.

Maybe you simply need a bigger amp! The dB defined.

Without getting too technical, the decibel, or "dB" is the unit of measure for "sound pressure level" or SPL. Because it is expressed on a logarithmic scale, it can be a little bit confusing. As an example, a doubling of volume is not equal to twice the number of dB. Studies have shown that most humans cannot detect a difference of volume of less than 3dB. Yet it takes twice the *power* for a 3dB increase in SPL. Now here's the thing. To double the *volume* (an increase of 10dB), **we need ten times the power!** Hypothetically, your 5W amp might make 93db – which is freakishly loud in your living room. But to hear on stage with a typical live band you need about 103dB– that's an increase of 10dB. Therefore – going from a 5W amp to a 15W amp isn't going to do the job. **You need 50W.** (Yes, everyone should play more quietly – there's no reason to play that loud, etc. etc. If you have your own band you have some control. If you don't, you don't. I have a sound pressure level meter and I have used it in live settings. The above numbers are indeed very realistic.)

More Speaker, too

As power goes up, so should speaker surface area. Trying to push 50 W through a single 12 inch speaker usually results in harshness of tone and less air movement. More speakers, given sufficient power, increase volume because they move more air. Doubling the number of speakers produces about a 2dB increase in SPL for the same power. Compare both power **and** the surface area of the speakers on different amps. A Fender Blues Jr, is a 15 watt amp with a single 12" speaker – if you do your "Pi r^2" you'll know it has 113 sq. in. of speaker surface area. My Sonny Jr. Cruncher is about 35 watts, and has a 12" and two 8" speakers. That's 213.5 sq. in of speaker surface area – more than twice as much power and almost twice as much speaker area. I can get the volume on that amp to 9 with no feedback because it is built to be harp friendly. But even that isn't loud enough at some of the jams I go to. If I want to be heard at a jam, I bring out the big guns. My Sonny Junior "Avenger" amp has about 50 watts, and has a 12" speaker, a 10 inch speakers and two 8" speakers. A little more power, but another big jump in speaker surface area - 292 sq. inches to be exact. The difference is huge. Room filling volume, and again, completely manageable feedback!

For any given amount of power, *gain* is the most important variable in taming an amp's tendency to feed back. Remember, an amp is only so loud. No 5 watt, 8" speaker amp will cut through at a jam unless it is mic'd. The best weapon of all is a more powerful amp with more speaker area. But a typical guitar amp *can* be made much more harp friendly by reducing its gain, so let's talk about that for a moment.

Reducing feedback by reducing gain - amplifier tube substitution

First, let's define gain. Gain is *not* the same as volume. Gain is the *slope* of the line mapping input volume to output volume. Here's an analogy. Imagine you have a Mazda Miata with a 500HP V8 up front. Now imagine the gas pedal only moves 1/2". *That* is what too much gain feels like. Idle is idle, full throttle is full throttle - but the engine is so hard to manage *in between*, you have to drive very slowly to be safe. Now imagine the same car but with 5" of gas pedal travel. Idle is still idle, full throttle is still full throttle, but in between it is much easier to sneak up on that power smoothly. You will get around the racetrack much faster with this setup.

This is completely analogous to how gain affects an amp. Reducing the gain is like adding throttle pedal travel. It doesn't increase power, but it **does** change the nature of feedback. Instead of leaping out of the amp at the slightest touch, it comes on more gently and progressively, allowing you to control it with your technique and volume control. For that reason you can actually use more of the amp's available power without feedback.

Unfortunately you can't easily adjust the gain of most solid state amplifiers. For those, a more expensive solution in the form of an anti-feedback pedal such as the "Mojo Pad" (http://www.lonewolfblues.com/store/harp-effects/mojo-pad), "Squeal Killer" (http://thesquealkiller.com), or the Kinder Anti-Feedback + (http://www.kinder-instruments.com/afb+.htm) can be very effective.

However with tube amps we can simply substitute lower gain tubes for higher gain ones in the amp's preamp section – **and this can instantly make an amp more harp friendly.** The most popular preamp tubes are the 12A# series. A typical Fender guitar amp has three tubes in the pre-amp section, and all are 12AX7's as the amp comes from the factory, set up for electric guitar.

Here are the gain factors of the 12AX7 and its compatible cousins:

GAIN FACTOR 100: 12AX7 - aka ECC83, 7025, ECC803, E83CC, 6681
GAIN FACTOR 70: 5751
GAIN FACTOR 60: 12AT7 - aka ECC81, 6201, 6679
GAIN FACTOR 45: 12AY7 - aka 6072

GAIN FACTOR 41: 12AV7 - aka 5965

GAIN FACTOR 19: 12AU7 - aka ECC82, 5963, 5814, 6189

As you can see, three 12AX7's creates a huge amount of gain. On an amp like this, you will likely not get the "normal" channel volume past 2 before feedback. If your amp has an "overdrive" channel it will be completely unusable. There are way too many combinations and permutations to describe here, but here is one recipe for success: On a Fender Bassman or Blues Jr amp with three preamp tubes, replace the two 12AX7's closer to the amp's center (named "V2" and "V3") with 12AU7's. This should make a significant difference. If it isn't enough, replace the outer one with a 12AT7 or 5751. You should now be right in the ballpark and grasp the concept - continue to experiment and pick what you like.

How to be heard

Although reducing the gain of your amp makes it more manageable, it will not perform miracles. A Blues Jr sounds ear-splittingly loud in your living room. Yet you get on stage at a jam and wonder why it can't be heard. So you turn it up until it is feeding back and conclude that you have a feedback problem. But you don't. You have an amplifier problem. Said another way, **just because your amp doesn't feed back easily doesn't mean it is loud.**

Mic it

One solution is to mic the amp or use its line out feature (if it has one) to connect the amp to the PA system in use. This is effective and allows much more volume for the audience. Small amps can have outstanding tone – and they're less expensive. There is no free lunch, however. The volume "out front" may be sufficient but if you can't hear yourself, or the band can't hear you, it is very difficult to play well. And small amps can easily be drowned out when stage volumes are high. The "normal" solution to helping people hear themselves on stage is to add some of what the audience is hearing through the PA into the "monitors" – those speakers on the floor at the front of the stage, facing the musicians. Unfortunately you'll find that having much of the harmonica sound in the monitors is a feedback nightmare. It is best to be able to hear your amp itself.

If you're playing at jams, you may not have the option to mic your amp at all.

Line Out

Some amps have a "line out" circuit – a jack that provides the amp's audio at an appropriate level to plug into another amp or the PA. The idea is the same as mic'ing an amp – just without the mic. It has the advantage of being easier to connect to an amp on stage, and eliminates the common problem of someone kicking a floor mic out of position. Good line out circuits are driven from the amp's speaker circuit and as such can have a very good representation of the amp's tone.

Other feedback management tips:

Avoid "overdrive" amplifier channels or pedals: generally speaking these work by adding lots of gain and will be feedback nightmares. If you're using a PA system and your amplifier is mic'd, tell the sound guy to take you out of the monitors completely. If you can't hear yourself on stage, and getting closer to (or further from) your amp doesn't help, have him sneak a tiny bit back in. Change where you stand. Turn the treble or midrange down. Use an "anti-feedback" pedal. If none of that cures your feedback problem, you are simply asking more of your amp than it can deliver. The bigger the amp and the more speakers it has, the more likely it is that you can get good volume before feedback, regardless of which mic you choose!

Volume controls

And finally, get a volume control! There is nothing worse than getting everything all set during sound check, and then encountering feedback during your performance. But this happens all the time, because room conditions change for a variety of reasons (the sound man included.) You certainly don't want to have to dive for your amp to adjust it, knowing the feedback problem will be worse when you get closer. A volume control at the mic is the perfect solution to this problem. You can instantly kill feedback from where you stand just by lowering the volume a bit. When you perform, you often can't be sure where the feedback is coming from. IF you hear feedback on stage (everyone else will too) – lower your volume control immediately. If the feedback persists, you get to throw your hands up with that "Hey - it isn't me!" expression.

Volume controls are useful tools even if feedback isn't an issue. Although any good player has good dynamic control over his instrument, there can still be a wider dynamic range in your playing and with the band than you can accommodate merely by playing softer. Being able to back the volume off right at the mic is really nice when you're comping behind a quiet passage in the music. You can also use a volume control to explore the tonal range of your amp, when otherwise you'd be firmly in feedback territory. An amp turned up to 10, with the mic's volume controlled by you to remain below the feedback threshold, will sound different than the same amp at 6 with the volume control on the mic all the way up.

Some players pooh-pooh volume controls. They're usually players who have simply never tried a good one! Practically every pro player I have seen uses a volume control. Charlie Musselwhite, Kim Wilson, Rick Estrin, Rod Piazza, Jason Ricci, Rob Paparozzi, Billy Branch, Curtis Salgado, Mark Hummel… the list just keeps on going. All of them use volume controls. And it isn't because they don't have excellent dynamic control.

What about Wireless?

Many players want to go wireless – fun, because you can go out in the audience and play, dance up on the bar, or merely be free of that cumbersome cable. A wireless system always consists of two parts. The transmitter stays with you, connects to your microphone and sends the signal out into the air using radio waves. The receiver is located near and connected to your amp. There are many types of wireless systems available to us. None of them are totally transparent – they all tend to "compress" the sound to some degree, which you may or may not like. As a rule, you usually get what you pay for. But there are some practical considerations.

Some wireless transmitters are small and physically connect to your mic, making a single self-contained unit. Others need a cable that connects to a "belt pack." The belt pack systems are often of very good quality and fairly economical, but they have a drawback in that you cannot simply put your mic down and move away. Because I double on saxophone in my band, I need to switch instruments quickly and so I have to be able to just lay the mic down. With a belt-pack system I would still be tethered to the mic. Some belt-pack users will simply shove the mic in their pocket – if this is okay with you don't rule out belt pack systems.

Of the self-contained systems, there are two flavors. The "guitar bug" is a device designed for (guess what) an electric guitar. It has a 1/4" plug on it – and can be used with a high impedance mic with a 1/4" jack. An example of this is the Samson Airline 77 system with the AF1 transmitter.

The other flavor is designed to plug into the end of a vocal mic. It therefore has an XLR connector and is barrel shaped. A good example of this kind of system is the Samson CR77 with AX1 transmitter. Although the XLR connector might suggest this transmitter would only work with low impedance microphones, it turns out this transmitter is perfectly happy working with a high impedance microphone, provided that the microphone's XLR connector is wired "Pin 2 Hot." BlowsMeAway Productions sells adapters for microphones with screw on connectors, with and without built-in volume control, that provide exactly this connection so that these mics can be used with this system.

Regardless of the wireless system you choose, it is important to know that they must be properly set up if your goal is a "cable-equivalent" sound. Set up improperly they will be very different.

Set up correctly, they can come very close to a cabled sound. BlowsMeAway provides wireless systems specifically for harp players, and will set your system up for your microphone and amplifier. See the http://www.blowsmeaway.com/wireless.html page for more information.

Important to users in the United States: Under an FCC rule anyone who uses a wireless microphone that operates in the 700 MHz Band, and that means almost every wireless system sold prior to late 2009, is supposed to have stopped using it no later than June 12, 2010. This action helps complete an important component of the DTV Transition by clearing the 700 MHz band to enable the rollout of communications services for public safety and the deployment of next generation 4G wireless devices for consumers.

For further information, please visit the website at http://www.fcc.gov/cgb/wirelessmicrophones

Choosing a mic builder

I have worked on many mics that others have worked on before me and I can tell you that mic builders are NOT all created equal! There are a few very, very good ones out there. And there are some guys who rave about their mics but do really poor quality work. I urge you to consider the following:

1. Choose a builder who is easy to communicate with! If you can't get your emails or phone calls answered in a day or two you're dealing with the wrong guy.

2. Check the builder's reputation. Google him. Ask online. If someone has had a bad experience they may not choose to air their laundry in public, but they'll send you a private email. I have gone so far as to conduct a web-based customer satisfaction survey and have posted the results on my web site.

3. See who the builder's customers are. If pros are willing to associate their name with the builder, that's a strong recommendation.

4. Can they play? I mean really play? A mic builder doesn't have to be Charlie, Kim or Rod, but if he can't at least produce great tone and demonstrate good microphone technique then he probably doesn't "get it." If he doesn't perform, he may not understand what's important, not just about tone, but about compatibility, reliability, feedback control, etc. – experience that comes from the performance environment. And if he can play, I guarantee there are videos and/or sound files on the web and/or CD's to prove it.

Conclusion

This has been a rollicking stream-of-consciousness core dump on my part. I hope it has been helpful. There are many subjects left untouched. In the end only one thing matters, and that is your personal satisfaction. You can't know enough when you buy your first mic. Only when you buy your second can you understand how it differs from the first. The more you try, the better your technique becomes, the more situations you play in – the wiser you will become and the better you will be able to hear the differences. Still, you can avoid wasting a lot of money by choosing your sources wisely, and then trusting their advice.

Feel free to contact me at BlowsMeAway Productions by email – greg@blowsmeaway.com - any time if you have a question. That's what I'm here for!