# Harp Amplification

If you think there is a single best way to amplify your harp, then you're going to be dissatisfied. There is no best way to do this. Unfortunately, amplifying acoustic instruments is devastatingly difficult and fraught with problems, producing poor results for much of the time. However, with care and attention to detail, good results can be produced, although not without a certain amount of expense.

Firstly, what is sound? Well – without too much detail, sound is movement of air molecules. All acoustic instruments just cause the movement of air molecules in a very precise way so as to produce at our ears the illusion of a specific sound or series of sounds. And if the instrument causes a large amount of air molecule movement, we hear a loud sound, and vice versa. So, because some instruments only move a small amount of air, we need amplification, to be accurate second stage amplification, because any acoustic instrument is already an amplifier. All acoustic stringed instruments amplify string vibrations producing a louder sound than if the string were just plucked or excited without the instrument. We need to understand how a harp amplifies its sound, before we can amplify the actual harp. When you pluck a string, it vibrates and that makes a sound. Without a sound box, the sound would be pretty weedy! However, as the string vibrates, it pulls the soundboard up and then releases it, then pulls it up again, in time with the vibrations from side to side. So, in turn, the soundboard vibrates up and down. A string only moves a small quantity of air when it vibrates so the sound is thin. The soundboard has a much greater surface area and so when it vibrates, it moves a much bigger amount of air, and so amplifies the sound. Now, if you imagine having a soundboard suspended by itself, without any box around it and supporting it, but with strings coming out of it, and you plucked a string, you can see that you move sound above the soundboard. However, you also move an equal amount of sound below the soundboard so you get sound from the back of the soundboard as well. You can directly compare this to a loudspeaker unit, with the cone of the loudspeaker being analogous to the soundboard. Now, you probably know that if you play a loudspeaker unit by itself, without any baffle or enclosure, in other words not in a box, you get a poor sound. This is because the sound from the back and the sound from the front are exactly out of phase with each other. They cancel each other out. So you mount the speaker on a baffle or in an enclosure in order to separate the opposing sounds, and in this way prevent them cancelling each other. Similarly, the sound box of the harp performs the same function – it separates the out of phase sound from the back of the soundboard from the sound from the front of the board. So you get a lot of sound instead of almost no sound.

If this was all it did, then all similar instruments would be much of a muchness – and we wouldn't have good instruments, bad instruments, exciting instruments, boring instruments etc. etc. But it also does something else; at the same time as the instrument's body is amplifying the signal, it is colouring it with the particular instrument's qualities – the individual characteristics that make the instrument unique.

But increasingly in modern performance most acoustic instruments just don't make enough sound. And the harp is one of the quietest stringed instruments. If you just want to play it in your living room, then it's fine. If you're playing professionally to a small listening audience, then it's fine. But if you're in a big auditorium, playing at a wedding (where maybe everyone isn't quietly listening to you), or even just playing in the company of other musicians with other instruments (mostly louder than yours!) then it just isn't fine at all.

So how do you get around this problem? Put simply –you just need more sound. It isn't simple. There are a variety of solutions, suitable for a variety of situations, costing various amounts of money but never very cheap!

The first thing you need to think about is why do you need the amplification? Is it a small amount of reinforcement at small recitals? Is it reinforcement at fairly noisy events like weddings? Is it because you play in a band and can't be heard amongst the other members? Do you need reinforcement, or do you need actual full-blown amplification? All of these situations demand different approaches. Sometimes the answer doesn't involve electrics at all – Rhona MacKay, when she started to play in the Whistlebinkies, got a bigger harp because the Whistlebinkies are a solidly acoustic group.

However, for most the solution is amplification. The problem is the amplification in our acoustic instrument was responsible for the unique qualities we like in our instrument. But now we want amplification that doesn't contribute any additional characteristics to the overall equation. We want to hear our instrument's sound, but louder. That's all. And that's difficult. Just as the instrument introduces colouration, so does electric amplification. We are essentially trying to amplify an already amplified situation.

The best way to get more volume is always from an electric instrument – one that has been designed from first principles to be electric. But that isn't always what people want.

So we want to amplify our existing instruments, at least until Starfish get round to producing the first electric harp that sounds and plays like an acoustic harp!

**What exactly is amplification?**

Well – the amplification chain has three main components in it: a pickup system of some sort, an electrical amplifier, an output system – loudspeakers, and signal cables connecting them.

The pickup system is responsible for collecting either sound information in the air, or vibrational information from the instrument, converting it to electrical energy, and transmitting it via cables to the amplifier, which increases the amount of electrical energy, and sends it to the loudspeaker again through more cables. The loudspeaker converts the electrical energy back to sound energy and transmits it into the room. And each of these links in the chain is trying to have an effect on the overall result.

We need to consider each of these to decide which will be most suitable for our specific situation.

**Different types of pickup**

There are broadly two types of pickup system: microphones and contact transducers or similar devices.

All microphones convert sound energy into electrical energy, but there are many ways of doing this. In other words, mics are sensitive to sound in the air. Fortunately for us, the vast majority of mics used in amplifying musical instruments are either capacitor (condenser or electrostatic) or dynamic (electromagnetic) models. Both types use a moving diaphragm to capture the sound, but use a different electrical principle for converting it into an electrical signal. The efficiency of this conversion is important, because the amount of acoustic energy produced by either voices or musical instruments is small.

Dynamic mics are the ones most used for live music – they are the ones that look like a mesh ball on a stick. They're cheaper than capacitor mics, don't need a power supply, and are hardwearing. They are suitable for working with relatively loud signals that don't contain much high-frequency material. In other words, fine for us.

Capacitor mics produce such a small signal that they need a built in pre-amp to bring the signal up to useable levels. This and the fact that they require a voltage to work at all means that they need power supplies. This is normally done using 'phantom power' which sounds a bit strange but will be explained later.

There are other types of mic such as electret and back-electret but they are generally just variations on a theme.

One of the problems with any pick up system is its frequency response. You might expect it to be linear – in other words equally responsive throughout the spectrum but that is really never the case, sometimes on purpose and sometimes by accident!

Most mics have a deliberate low-frequency 'roll-off' – in other words they're less sensitive to low sounds. This is on purpose, because if not, mics would amplify knocks or breathing noises for instance, with annoying results. Equally some mics have an accentuated response in certain areas, for instance to help make vocals more intelligible.

There is also a thing called the proximity effect, which you'll have seen often when singers appear to be almost swallowing their microphone. If a mic is very close to the signal source, the bass in the signal goes up enormously. It's a problem or a benefit depending on what you do with it!

One more thing that needs to be considered when using mics is their directionality. Omni pattern mics tend to be most accurate but may pick up signals not intended for them, so often directional mics need to be used. These are known as cardioid mics.

Often, because of the spread of strings, players use more than one mic to get a balanced sound. Sometimes people struggle greatly with this. It has been reported that while recording her first CD at a studio, a well known harp player finished up with 13 mics around her – it was a very intimidating experience. She didn't go back for the second CD.

Properly set up, a mic system produces the very best of amplification, but it's not cheap. It's the only system that can give the impression of reality that in an ideal world all players would demand. But it does suffer from volume level limitations caused by feedback.

You can use two different mic systems: mics on stands and mics which are attached in some way to the harp. With stand mics, an acceptable sound can often be got from one mic pointed at a spot approximately two thirds of the way down the soundboard. Try to place the mic as close as possible, bearing in mind that the player needs to move the harp on and off their shoulder.

However, two mics are much better. Place the first mic the same as in the solo case, but just a little further down the soundboard. The second mic should be placed below the right hand pointing at the strings – making sure it doesn't get in the way of the player! You may need to adjust the Equalisation between the two mics. This will give a combination of full sound from the soundboard, with that elusive quality of string movement. The kind of amplification you don't notice.

Microphones inside the harp in general don't work well, as there is too much resonance – there's a lot going on inside a harp and expecting the mic to pick out the musical bits is too much!

To set the volume, there is a natural point of sound level in any room when the sound is natural, good to listen to, and yet still loud enough. This is the point where you become unaware of the presence of the sound system. In most reasonable sized situations this level should be set for about the third row of seating in the audience. We are talking ideal scenarios here, but if you do get it right, it's worth it.

Apart from larger mics on stands, there are available smaller cardioid capacitor mics which attach directly to the harp. Cardioid mics are directional, in other words they pick up sound only from the direction they are facing. They have the advantage that you can move the harp without the mic becoming out of place. One of the best is the one made by Applied Microphone Technology, an American company. They advise that the mic be placed by a sound hole, but we certainly don't – we place the mic over the soundboard approximately halfway down, and facing slightly more toward the strings than the soundboard. This works well.

There are many types of contact device – these are the second type of pickup – those which are only sensitive to vibrations within the instrument. The most common are those based on piezo technology. All piezo transducers work by vibrations causing the deformation of a piezo crystal or film, with the resultant generation of electrical energy. They are of varying quality and efficiency, with the lowest output type requiring preamping.

Cheaper types of contact transducer include various bugs or 'hot-spot' type discs. The little bug used with your electronic tuner is a good example of the cheaper type of these.

All of these pickups tend to be applied to the body of the harp either on the soundboard in front, or in the back somewhere. They are stuck on with double-sided tape and although in theory they can be removed, it's often easier to leave them in place. A certain amount of experimentation is usually necessary to find the best spot to place them. As is common with mics, often more than one is used to amplify the spread of strings.

Another device that exists is a strip that is built into the back of the lower string bar during the manufacture of the harp. They can actually be fitted after as a retrofit, but this is difficult and expensive. It produces a quite dramatic sound but unfortunately overly coloured and not very pleasing.

Some of these systems produce a very low output which requires preamping – usually with a small unit that can fit into the back of the harp and is battery powered.

The disadvantages of these devices can be a compressed or 'wooden' type of sound, non-linear sound quality over a range of dynamics and frequencies, and high sensitivity to knocks and other extraneous sounds caused by touching the instrument.

Generally you can attain higher sound levels than with mics, but you may not want to because of the increasing lack of quality with higher levels.

One point worth mentioning is that it is absolutely irrelevant whether you place the contact transducer on the soundboard inside the box or outside on the front of the soundboard. The transducer is only sensitive to vibrations within the soundboard and these are exactly the same when measured from either side. It's where it is placed that is important.

There is a system available for guitars that incorporates both a mic for accuracy of sound, and a piezo contact transducer for higher sound level – this is regarded as a very good system, but unfortunately it's not optimised for the harp so doesn't work very well. It would be the ideal though if a version were to be produced specifically for the harp.

**Amplifiers**

All amplifiers consist of two main parts – a preamplifier and a power amplifier. The preamp part merely boosts the often rather weedy signal from the pickup to what is called line level while at the same time any equalisation or tonal modification is done. The resultant line level signal is then amplified by the power amplifier to a level that can produce sound from loudspeakers.

There are no amplifiers made specifically for amplifying harps. The market is too small. Therefore we have to fall back on using amps that have been designed for other purposes. The question is – which are best for our purpose and why?

Briefly, amps fall into four categories, mostly based around guitars.

a) Acoustic amplifiers

These are designed to reproduce exactly the sound of an acoustic instrument without colouring the sound. This is called transparent. Unfortunately it's difficult and difficult means expensive. Often these amps can be very good for amplifying harps.

b) Electric Guitar amplifiers

There are more amps in this category than all the others put together. With a few exceptions, most of these are not suitable for amplifying harps.

c) Bass Guitar amplifiers

Surprisingly, many of these amps can be very good for amplifying harps, although the cheaper models can suffer from a lack of top end.

d) Keyboard amplifiers

Many keyboard amps can also be very suitable for amping harps, although not all.

**Different amplification situations**

There are many types of amplifier and speaker combination.

a) PA or submixer

In situations where the harpist is playing in a band, the amplification is usually done with a PA system, which consists of a mixer-amplifier and usually stereo speakers on stands. The mixer-amp will have multiple inputs, one of which can be used by the harp pickup or mic. A sound engineer usually controls the ‘mix’ of inputs from the band members, although a band member can do this.

b) Combo amplifier

Combos are so-called because they combine the amplifier and speakers into one cabinet. They are more convenient than a PA and often sound good because they are designed as a single unit. Individual players usually favour this approach, although once again individual band members may prefer this approach as well, as they are more in personal control of their sound.

c) Heads and speaker cabs

In this configuration, the amp is separate from the speaker cabinet and is called the head. It will normally sit on top of the speaker cabinet. This gives the player more control over matching an amp with different cabinets – for instance, two different configurations of cab would be 1x15 which contains one 15 inch speaker and 4x10 obviously 4 10 inch speakers. This gives much greater attack and results in more clarity and presence. Thus it would be much better for a harpist. These tend to be used in larger venues than combo amps.

Most amplification has similar controls and these are the subject of a certain amount of mystery.

**Channel gain:** this controls the input level of the signal from the pickup system. Generally it is best to run it up as much as possible before causing distortion. Amps operate best with a strong signal and there will be less general noise as well.

**Equalisation (or EQ):** EQ refers to boosting or cutting frequencies to give the balance desired by the player. It should ideally be used to compensate for room inadequacies rather than be a way to try to get an acceptable sound from a rubbish pickup. There’re several types of EQ:

a) Shelving – the simplest, usually three knobs or sliders called treble, middle and bass. It’s called shelving because each control acts purely on predetermined frequencies.

b) Graphic EQs consist of sliders that control various frequency bands. Set central, they neither boost nor cut.

c) Parametric EQ allows the user to select a frequency or frequency band and boost or cut just those selected frequencies. It is much more precise than the other types of control, and is often used to sort out problems. There may be a Feedback control on the amp, which is essentially a parametric EQ that is only capable of cutting a problem frequency, not boosting it.

Volume: this controls the overall loudness of the amp.

**Reverb:** this is a useful effect. It simulates the effect produced in large echoey rooms or halls. Used carefully it adds presence to an instrument without giving it a processed sound. Bowed instrument players simulate it acoustically when it’s called vibrato.

**Effects:** this will either control outside effects (plugged into a loop) or the amps internal effects. It’s probably not going to be used much by the average harp player, although you never know!

Usually the amp will have a standard ¼ inch or 6.5mm jack plug connection for the input. This will be unbalanced. Additionally there may be a balanced XLR input. This will be used in big stage situations where long cable runs are necessary and will incorporate phantom power.

This brings us to the next link in the chain: cables.

**Cables:**

Most often the original mic signal is carried from the mic to the preamp by a balanced XLR cable. The line level signal can be carried by either a balanced XLR cable or an unbalanced cable with jack connectors. An XLR cable is composed of two thin insulated wires twisted around each other, with a braided sheath surrounding them, which is grounded. The signal is carried in the two cables and any noise is cancelled out because it is common to both cables. No signal is carried in the shield or ground connection. So they can be as long as you like.

A cable with jack connectors on each end is unbalanced – it consists of just one thin wire that is insulated from the sheath, which acts as the second signal carrier, and also the ground. Obviously any induced noise in either conductor will not be cancelled by similar noise in the other one and will be carried with the signal into equipment. Therefore unbalanced cables should be kept as short a possible – certainly under 8 metres.

So we can have three kinds of cable:

a) Balanced microphone cables: mic to preamp or DI box. These cables have low voltage, low distortion and a low hum level.

b) Balanced line level cables: these are interconnects between preamps, DI Boxes, EQ boxes, effects units and amplifiers. Again they can be any length but they are usually short as they just interconnect these boxes.

c) Unbalanced pick up cables: Used from the instrument to the preamp or DI Box. Very susceptible to noise and should be kept as short as possible.

**Set up Issues**

So we've discussed each of the links in our amplification chain – now we have to connect them together.

At this stage we need to introduce another problem: impedance. Unfortunately it's an issue that isn't properly understood even by some studio engineers, although in practice it's pretty simple if you follow some basic rules. The term impedance may be thought of as AC resistance, or resistance under signal conditions. Like other types of resistance it's measured in ohms. In audio equipment, and signal cables particularly, impedance will vary with frequency. And as we know, frequency varies most of the time when you're playing instruments. If a pickup has an output impedance that is higher than the input impedance of the amplification system we are in for disaster. And the level of disaster will increase with the length of cables used. It's like trying to get something that's just too big through a bottleneck – effectively something will get through but not all of it and the sound will be strangulated because it'll lose some of its frequencies.

Often you will find that amps have dual inputs, with one labelled high and the other low. Don't be fooled by this! The 'high' is only high in relation to the other one, which is a low-impedance microphone type input, and it will still be around ten times too low to accept the input from the average piezo device, which has an impedance of around 1megohm.

The solution is to use a DI box. DI stands for Direct Injection.

A DI Box offers the instrument a high-impedance, low-capacitance input, giving maximum signal level and a flat frequency response, and it provides a low output impedance that isn't affected by the mixer or amp input load.

Let's examine the different signals we need to amplify:

a) Low level balanced microphone signal:

Pretty well all mics are covered by this description.

Signal level is small, but impedance is very low also, around 600ohms, so very long cables can be used, so long as they are well shielded.

b) Unbalanced instrument level:

This covers most piezo or contact transducers. These signals are the ones with the high impedance – often, they not always they tend to be small, so if they are sent down any appreciable cable length, noise will be introduced and quality compromised. Thus a DI box should be used, which should balance the signal, reduce its impedance, and allow longer cables to be used.

Often, if the signal is low, then a preamp will be used, in which case the DI box is usually unnecessary, as the preamp will accept the high impedance input of the pickup and provide the necessary low impedance out put at the other end. Often the preamp will also possess a volume control and a rudimentary graphics equaliser – usually of the shelving type – typically with bass and treble controls, but sometimes with a middle control as well.

Good DI boxes also provide a 'loop-through' facility so that a signal can be passed on often to the musician's own amplifier, while the DI'ed signal goes on to the mixing desk.

There are two main types of DI box – passive and active. Passive boxes do not have any form of amplification built in, don't need batteries or phantom power, and are cheaper. But they don't improve weak signals, often can't cope with the ultra high input impedances of some pickups, and can 'load' the pickup in a way that can dramatically affect the instrument's performance – and not in a good way! Most passive DI's can only manage around 100k.

Active DI boxes are the ones to go for. The big difference is that they incorporate a buffering amplifier which isolates the instrument's pick up from the rest of the DI technology. This box usually gives trouble free results and equips you to plug into any kind of amp, PA or mixer. But you need batteries or phantom power.

One point to consider here, if you have the facility to connect to anything around – whether it be your friends equipment, the equipment at the venue, or any other kind of unknown kit, is be careful! Connecting to unknown equipment is potentially dangerous. If something looks even slightly dodgy – don't plug your DI, preamp, or mic into it! At best expensive kit can be destroyed and at worst, people can even be killed. And here, if your DI is good enough it would provide you with some protection, by using its transformer to electrically isolate the musician's equipment and the mixer or PA.

One other common, but less dangerous problem is that of ground or earth loops. These are caused when a loop is formed with the connecting audio lead between the earth in your mixer's mains plug and the mains plug of your musician's own back-line amplifier, or the instrument's main plug if it has one ( like keyboards, for instance). The solution is to either disconnect the screen at one end of the audio cable, or use the 'ground lift' switch on the DI box if it has one. Both of these isolate the two earths, and break the loop.

Don't forget that in almost all cases, using pickups on acoustic instruments such as harps, will inherently compromise the sound, so this is usually only the solution if it allows greater freedom of movement, reduces mic clutter, or is cheaper than a pure microphone solution.

All acoustic instruments are designed to be heard at some distance, and the sound quality is almost always inferior when reproduced through a pickup mounted on the body of the instrument. Corrective EQ is usually necessary in these cases.

Feedback is a universal problem when amplifying acoustic instruments. It can be caused in practice in many situations, but essentially it is a problem of pure physics. When a signal to a microphone is amplified and used to drive a loudspeaker, some of the sound produced necessarily returns to the microphone and is amplified again. This is called feedback. And it is feedback that is almost always the factor that limits the maximum amplification from a sound system.

If you speak one word into a microphone at a level of 80 decibels – decibel is a measure of sound pressure, or simply sound level – and the loudspeaker amplifies the sound and manages to send the word back into the microphone at the same 80- decibel sound level, then you can go home. The sound system will repeat the sound all day – chasing its own tail. Similarly, when the gain on the amplifier is turned too high, the output from the speaker changes to a very loud usually fairly high-pitched sound, which is the result of too much feedback, but the signal being amplified is drowned, by a single pitch at the frequency which is amplified most by the sound system and room acoustics combination. This is also picturesquely and much more realistically sometimes called howl round.

Various fairly obvious steps can be taken to try to minimise this –

Move the speakers further from the microphone. Often, you don't achieve much by this as the reverberant sound field caused within the auditorium from reflections eventually causes as much feedback as the direct sound field does or did.

Move the loudspeaker closer to the listener or, we hope, listeners. You can also add extra speakers that are closer to the listener. Some times this can produce imaging problems, because of the sound from the nearest speaker reaching the listener before the sound from the visible source at the front of the auditorium.

You can move the instrument closer to the microphone.

You can move the instrument closer to the listener.

Additionally, as has been mentioned earlier, use of directional mics helps and use of directional speakers helps.

Careful EQ to screen out room resonances can help, and there are electronic devices called notch filters that also help. They are also called feedback suppressors.

So – mics have limitations! How about pickups?

Often people think that by DI'ing a pickup, they will avoid feedback issues. This is generally true in the case of electric and electronic instruments, but definitely not the case with pickups on acoustic instruments. The harp's soundboard amplifies string vibrations and radiates sound out to the audience. But the same soundboard is just as capable of collecting the sound from the amplifier's loudspeakers and sending it back into the pickup. If you can't use a microphone in front of a harp because of feedback, using a pickup will only be marginally better, at best!

As mentioned earlier phantom power is a method of supplying power through a balanced cable back from a mixer or amplifier to a microphone or DI box. The power is used by the pre-amplifier within the microphone or the DI box. Phantom power is supplied most usually at a nominal 48 Volts, so it's higher and steadier than using batteries in a device and is always a better idea if it's present. But often batteries have to be used – cabs often don't have phantom power and often small preamps will only work with certain battery sets.

If you use batteries, almost anything will do as the drain is very little, but you must replace them occasionally as a battery failure during a performance would be a bit of a disaster. Normal alkaline batteries are fine, but high power rechargeables like NiMH are not necessary. Often small inbuilt preamps take little calculator type batteries.

Hopefully this has given some insight into amplifying your harp. There's no substitute for real information, and the more you know, the better equipped you'll be to produce a good sound in any environment.

**Examples of Equipment:**

**Microphones:**

Neumann U87: this is the standard – a high quality studio capacitor mic. Cost: £1,500.

AKG C1000S: well used electret mic – much cheaper! Cost: around £150.

AKG C747: newer style hypercardioid mic – highly directional. Cost around £260.

AMT Harp microphone: Fixes to the harp. Cost around £300.

**Transducers:**

Starfish/Ashworth: inexpensive, simple solution. Cost: £180.

Accusound: similar cost, but uses inbuilt preamp. Cost: £100 - £180.

Headway soundboard strip: expensive, has to be built in during manufacture for best results. Cost: including fitting around £350-£400.

**Amplifiers:**

**Any AER model:**

Alpha – single channel, 40 watts. 10 x 10 x 9. 14 lbs. Cost: £465.

Compact 60 – Two channel, 60 watts. 13 x 10 x 9. 20 lbs. Cost: £595.

Compact Mobile – Same as Compact 60 but has inbuilt rechargeable power, giving 3-6 hours on one charge. Includes AER padded amplifier cover. 13 x 13 x 11. 35 lbs. Cost: £795.

Compact Classic Pro – this is a new amp designed to faithfully amplify classical instruments with no colouration. One channel acoustic amplifier for classical instruments. Pre amp section with 6 presets, by DIP switch addressable.Balanced mic-input stage, 48 volt phantom power. Three band tone controls and 32 bit AER classic effect processor. Cost: £899.

**Carlsbro:**

Carlsbro make a range of acoustic amps priced from £190 to over £300.

**Fender:**

Fender acoustic amps start at £270 and go up to over £600.

Most costs are approximate – you need to shop around if you can!