Sonic Studios™

DSM RECORDING TIPS **FAQ PAGE**

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Subject: Best way to mic piano for classical recording?

Newsgroups: rec.audio.pro Date: 1999/03/23

> I'm trying to figure out the best way to make a good classical recording of a grand piano. The piano is a full size Steinway, and the hall has excellent acoustics (Olin Arts Center at Bates College).
>
> I can record to either DAT or ADAT. I have a Great River MP-2MH preamp, and I have an assortment of microphones, but I'm not sure that I have the best mics for this occasion. I'm also not sure how to set up the microphones.
>
> I have (2) neumann TLM 103's, a shure sm81, an an audio technica AT4041. I'm guessing that a pair of small diaphram mics might be best, and I'm considering getting a pair of neumann KM184's. But maybe I need omnis, or figure 8's or something.
>
> Anyone have advice on what mics would be good for this application, and what sort of mic placement I should use?
>
> Thanks,
>
> Jeff Kew
> jeffkew@home.com

Subject: Re: Best way to mic piano for classical recording?

From: GuySonic (guysonic@aol.com) **Newsgroups: rec.audio.pro Date: 1999/03/23**

> Recording the sound of a grand depends greatly on capturing the ambient in a coherent manner that includes the direct sound from the instrument with the ambient mix or sound of the hall. Without including the ambient in a natural coherent manner, the recording is going to sound more like a tinted two-dimensional 'piano wallpaper' version than the real thing.

Therefore, use of an ambient stereo mic and method that records like you hear sound will give consistent satisfactory results suitable for speaker and headphones and sound that's 'virtual' to the listener, identical to hearing that grand in that hall at the mic position.

The mic position is any place fairly close to the piano that gives a good mix of piano + ambient. Too close will make the piano far bigger than life (not a bad sound if working in a poor sounding hall), while too distant will simply sound like too much hall and not enough piano that sounds a bit too distant; this is simple enough.

Just listening 'with both ears' will give you what you want in the recording using an HRTF baffled omni pair. No other mic method works as easily or consistently to record what you're hearing.

Baffled omni (either Earthworks Matched Pair ..-30K or Sonic Studios DSM-6S/EH with PA-10PFC powering) will record the full ambient and natural sound of this instrument (and the ambient) and provide 'as you hear it' mic placement not possible with a 'finger in one ear' or stereo mic suggestions presented so far.

Only one HRTF baffle design exists that will do this job reliably....... the LiteGUY (reviewed by Fletcher in this NG and in an upcoming review in Pro Audio Review by Russ Long...... should print in May or June) or the DSM-GUY. Both exclusive from Sonic Studios until the 'others' catch up on understanding the acoustics.

The DSM mics are shown on my web site (WWW.SONICSTUDIOS.COM) with reviews.

DSM mic models are at: http://www.sonicstudios.com/dsm.htm

A discussion of the why's of the baffled omni method is at:
http://www.sonicstudios.com/multitrk.htm

DSM mic powering is discussed at: http://www.sonicstudios.com/pa_x.htm

DSM recorded piano on page: http://www.sonicstudios.com/mp3.htm

Best Regards in Sound & Music, Leonard Lombardo
Sonic Studios(tm) "Making Audio History With DSM(tm) Microphones"

<< Subject: Re: Best way to mic piano for classical recording?

From: GuySonic (guysonic@aol.com) Newsgroups: rec.audio.pro Date: 1999/03/24

From: "Jean-Marie MATHIEU" <jm.mathieu@wanadoo.fr>
Date: Wed, 24 Mar 1999 09:08:38 +0100
>Someone please elaborate on ORTF. Inquiring minds want to know. Also,
>what is the baffle technique mentioned?

You can find a description of this technique here :
http://www.josephson.com/tn5.html

>>

The HRTF Baffle method is discussed at: **http://www.sonicstudios.com/multitrk.htm**

The LiteGUY model baffle is shown at: **http://www.sonicstudios.com/liteguy.htm**

HRTF translates to 'Head Related Transfer Function' and is a general term for the effect of the head with and without the additional effects presented by the ears on the reception of omni microphones placed (in some manner) in close proximity to specific areas of a real persons head and/or ears. DSM translates to Dimensional Stereo Microphones.

HRTF is the modification of the normally flat and non-directional reception characteristic of a single omni microphone or multiples (two for stereo) of omni mics when placed near a real person's head or acoustically correct (LiteGUY) baffle. I know of some nearly successful attempts to provide this complex transfer function in a digital processing solution that have been attempted since DBX 'borrowed' a set of DSM mics for evaluation in 1986 (they 'nearly' wore them out in 3 months of continual laboratory use evaluating the HRTF effect and created a company division to 'try' to develop digital processes to recreate the HRTF effect in the electronic domain; this division later became an entirely different company whose efforts persisted after DBX closed operations).

For practical use with HRTF baffles, the omni mics must be as small as practical to avoid distortions associated with the mic body size interference on the acoustic energy; mic sizes of much less than 1/2" are best in terms of accuracy. Extended ruler-flat frequency response from below 10 cycles to beyond 25,000 is a decided advantage in an omni mic for HRTF baffle use. The DSM mics are specialized for this with many elecro/acoustical innovations not found in other very small microphones.

A BINAURAL microphone is a TYPE of HRTF baffled recording method that always includes mics positioned in or in close proximity of the 'ears'; like with Neumann, Aachen (heads), and Senn. 2002 series mics. The DSM HRTF (patented) method avoids the limitations presented by ear modification (like that with Binaural) HRTF by careful mic positioning in an adjacent area forward of the ears to the temple area of the head or related area on a baffle like the LiteGUY.

HRTF baffle recording has in more recent years been referred to as a 'Psycho-acoustic' method of recording (perhaps thanks to Ken Pohlman's writings) as it contains (by virtue of the HRTF mic modification process)'psycho-acoustic cues' that the ear-brain process recognizes as natural spatial sound perception information. These natural 'cues' are contained within psycho-acoustic or HRTF method type recordings like those produced by the DSM mics when personally worn or mounted on a baffle like the LiteGUY.

The recordings made with the DSM HRTF (patented) method are most consistent in providing 'as you heard it' recordings on both speakers and phones.

This of course makes setup far simpler because you can 'believe BOTH your ears' (for the first time in mic methodology history) to what the mic position sounds like to you regardless of anything else; and is most likely to sound the same over a variety of speaker systems to an even wider variety of listeners.

Learning to listen carefully with both ears is all you need to know about using this method, but for those 'old timers', just learning to listen at a mic position remains a true challenge of 'no small dimensions'. Pun intended!

> Best Regards in Sound & Music, Leonard Lombardo
> Sonic Studios(tm) "Making Audio History With DSM(tm) Microphones"

Subject: Re: stereo bar for spaced omni recording?

Message 16 in thread Newsgroups: rec.audio.pro Date: 1999/03/07

From: Scott Dorsey (kludge@netcom.com)

> In article <romain.920792965@kzsu> romain@kzsu.stanford.edu writes:
> >In an ideal world, what would the good folk of r.a.p use to capture the bottom octaves in proper balance with the other part of the spectrum?
> >Wouldn't most directional microphones tend to roll off the bass?

Most directional microphones turn into omnis down there. If you want to build a cardioid mike with good low end, you need to use a very tiny diaphragm and then you start running into S/N issues.

Personally, I use a pair of B&K 2615 measurement mikes, with the 4033 half-inch capsules. +/-3 dB points are at 2Hz and 35Hz. I set them up in a Jecklin disc configuration, which gives you good front imaging and still allows you to use omnis.

Gabe used to use a slightly different baffled omni configuration, the Schoeps Sphere. In comparison, the tonality was very similar although it required different placement than the Jecklin disc.

A friend of mine uses the Sennheiser MKH50 cardioids, but then has a pair of MKH20 omnis as outriggers to adjust the ambience. The Sennheiser RF cardioid mikes have pretty good low end response as cardioids go. I don't like the imaging, but he has the ability to adjust for the hall in post and the label he works for really likes that.

> None of is stuff is particularly cheap, I am sad to say. Getting the whole audible range is not a trivial matter.
> --scott
> --
> "C'est un Nagra. C'est suisse, et tres, tres precis."

Subject: Re: stereo bar for spaced omni recording?

From: GuySonic (guysonic@aol.com)

In article <kludgeF88976.G3K@netcom.com>, kludge@netcom.com (Scott Dorsey) writes:

>Subject: Re: stereo bar for spaced omni recording?
>From: kludge@netcom.com (Scott Dorsey)
>Date: Sun, 7 Mar 1999 13:52:17 GMT
>
>In article <romain.920792965@kzsu> romain@kzsu.stanford.edu writes:

>>In an ideal world, what would the good folk of r.a.p use to capture the bottom octaves in proper balance with the other part of the spectrum? Wouldn't most directional microphones tend to roll off the bass?
>
>Most directional microphones turn into omnis down there. If you want to build a cardioid mike with good low end, you need to use a very tiny diaphragm and then you start running into S/N issues.
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>--scott
>--
>"C'est un Nagra. C'est suisse, et tres, tres precis."
>
>

Hey Scott! You forgot to mention Sonic Studios LiteGUY with DSM microphones again! After Fletcher gave his personal experience account and approval in a recent post, it should be an option mentioned by now and is a fraction of the cost of many alternatives mentioned with performance that's at least equal to the best stereo microphones (regardless of cost) in terms of seamless imaging and frequency response.

Sound Clips of all kinds of music instruments and ambient recordings are available on my site listed below. Many magazine reviews and recording tips are also available there.

The DSM microphones are not nearly as critical of placement as the Jecklin Disk or other stereo microphones because of the natural mechanisms of sound reception that's being used with the DSM microphone. Just listening to the mix at the microphone postion is enough to get exactly that sound without worry.

This consistantly easy placement with Sonic Studios DSM mic is not possible with any other type of stereo microphone as they do not record sound like we tend to hear it. This is why Scott mentions that placement varies with the type of stereo mic used. Therefore, if the mic doesn't record sound like you are hearing it (a known fact for the microphones mentioned so far in this thread), proper placement is going to be much more difficult as the recorded results will sound vary different with every type of playback mode used (includes speakers of all types and headphones of all types).

The Jecklin Disk is a sort of baffle that has little to recommend as it is NOT a true HRTF (Head Related Transfer Function) design. Only two true HRTF baffles exist and both are available from Sonic Studios. The Schoeps Sphere is way off the mark in presenting a good baffle to spaced omni capsules and remains a mystery to me why a company as good as this (excellent mics for sure) created such an odd thing that only works well on rare occasion. I would mention others that are at least close to an accurate acoustical HRTF design, but no others exist so far. Only Sonic Studios has designed an accurate HRTF baffle and I can't tell you why I'm the only one who has it as the design was not that difficult.

(Look for an upcomming review from Russ Long in Pro Audio Review on using the LiteGUY baffle with the DSM microphone in studio sessions he recently has done; this will be on my site with permission after it's published)

Frequency response of the DSM microphones is ruler flat from 10 cycles to above 22,000 with significant response out to beyond 30,000. You couldn't ask for a more affordable and quality way to record ambient stereo sound. Large instruments like piano, organ, drum kit, choral, and orchestral are prime for using this type of microphone offered exclusively for over 13 years by my company.

Best Regards in Sound & Music Recording, Leonard Lombardo

Subject: Re: stereo bar for spaced omni recording?

From: Scott Dorsey (kludge@netcom.com)

Newsgroups: rec.audio.pro Date: 1999/03/07 Message 18 in thread

> In article <19990307153801.28255.00002247@ngol02.aol.com> guysonic@aol.com (GuySonic) writes:
> \>
> \>The Jecklin Disk is a sort of baffle that has little to recommend as it is NOT a true HRTF (Head Related Transfer Function) design. Only two true HRTF baffles exist and both are available from Sonic Studios. The Schoeps Sphere is way off the mark in presenting a good baffle to spaced omni capsules and remains a mystery to me why a company as good as this (excellent mics for sure) created such an odd thing that only works well on rare occasion. I would mention others that are at least close to an accurate acoustical HRTF design, but no others exist so far.

Neither the Jecklin Disk nor the Schoeps sphere are _supposed_ to model
the head in any way. That's not what they are for. They are both methods of introducing some directionality into a an omni microphone. In both cases, the output of the mike more closely relates to something like an ORTF setup than a headworn setup.

Ultimately, what a conventional stereophony system is aiming for is to recreate the wavefront in the studio. This is clearly not possible, since you have a three-dimensional wave and you can only reproduce it at two points. But you can get surprisingly close.

You could argue that since you're listening to the music with ears that are attached to your head that it's already going through a head-like transfer function once, and that you wouldn't want to run it through twice unless for some reason it's being bypassed in playback (like with headphones). I won't get into the argument, because indeed the system from Sonic Studios works pretty well. But it's an interesting argument to follow and you can take it both ways pretty well.
--scott
--
"C'est un Nagra. C'est suisse, et tres, tres precis."

Subject: Re: stereo bar for spaced omni recording?

From: GuySonic (guysonic@aol.com) Newsgroups: rec.audio.pro Date: 1999/03/08

In article <kludgeF88v9y.2rs@netcom.com>, kludge@netcom.com (Scott Dorsey) writes:

>Subject: Re: stereo bar for spaced omni recording?
>From: kludge@netcom.com (Scott Dorsey)
>Date: Sun, 7 Mar 1999 21:49:09 GMT
>
>Neither the Jecklin Disk nor the Schoeps sphere are _supposed_ to model the head in any way. That's not what they are for. They are both methods of introducing some directionality into a an omni microphone. In both cases, the output of the mike more closely relates to something like an ORTF setup than a headworn setup.
>
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>
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>--scott
>--
>"C'est un Nagra. C'est suisse, et tres, tres precis."
>

The twice HRTF process argument is a very valid issue especially with Binaural HRTF that uses the modification of the head_and_the_ears to modify the microphone reception.

Using the complexity of ear modification presents the problem of twice HRTF processing speaker playback woes that are well known. The DSM aproach is not so limited. I was granted a methodology patent on the process employed with the DSM HRTF method of using just the head_without_the_ears as the sole modifier of the omni microphones because the results are far more general purpose and very effective for ambient stereo recordings that are reproducible from most any stereo or surround speaker and all headphones, especially open air types.

The Binaural (in the ear) mic method is a very effective type of HRTF recording, but only for headphones, especially closed or 'in the ear' types where (as Scott mentioned) the twice processed limitation is circumvented by eliminating the head + ears for listening. Also, ears vary greatly with people, so a general ear modifier doesn't work for everyone. NOTE: Many commercially available Binaural microphones systems like the Neumann or Aachen Head use special filters to change the mic signal from true binaural to something else that works a bit better with the general population of headphone listeners with varied ear shapes.

However, head only HRTF modifiers (like the LiteGUY) are more effective as head size doesn't vary very much over the entire human population making this mic method consistent. Virtually everyone who hears the recording can relate naturally to the 3-D ambient sound cues without mental 'reconfiguration' of the sound reception cues.

Having the HRTF processing of just the head modifier (sans ears) seems not to be a problem (thankfully) as this sound is prime for listener's perceptions with using their own head + ears with virtually no confusion with the head-only HRTF double processing that does take place with speaker playback listening.

The ears do so much acoustic processing on their own, that the pre-HRTF processed recording just sounds like a live recording with coherent 3-D ambient sounds that give the recording depth. The feeling of depth is something that is largely missing from the other stereo microphone methods discussed and is one the main advantages that the DSM approach has over just 2-D stereo imaging that cannot capture the 'sound cues' like we need to hear them. Only by using a baffle like the LiteGUY can those 3-D wavefronts, that Scott previously mentioned, be recorded so that we can later hear them and re-associate them (a psycho-acoustical brain process) like we do when listening to the live sound at the chosen mic position. The depth is part of the surrounding space and needs be recorded with natural coherent mic methodology like the DSM. Using disks or spheres or whatever else you want that falls short of the true HRTF will not consistently allow making a coherent ambient 3-D surround sound recording.

Studio recording methods (as often discussed in this group) have mostly gotten along very well, although other times not so well (like with piano), without dealing with the real issues of ambient sound by using close mic, not-natural stereo mic, and multitrack methods with/without post-EFX processing. However, recording LARGE instruments that are the ENTIRE BUILDING, like an acoustic organ, really raise the question on how to record the sounds of an instrument that involves (or should involve) the entire 360-degree acoustic ambient.

This can only be done successfully, in my knowledge, by recording all the sound in a 3-D 'coherent' surround-sound manner. While there are many approaches to doing this, only one mic method (DSM...HRTF) really delivers consistent performance regardless of working ambient (stage width, distance, ambient reflections, and so on) by recording in a very natural (coherent) manner that replicates own natural stereo reception mechanisms and consistently fulfills the need of the listener (our 'end customer') to effortlessly re-live the acoustic experience from the resulting recording.

It's not really that high tech, just very 'high natural'. And until I get greedy (or perhaps just more realistic) on my pricing, the DSM system is a bargain you can't refuse to take me up on if you want to bring that sound back fully alive virtually (pun intended) every time.

Best Regards in Sound & Music, Leonard Lombardo

Subject: Re: Hail Curtis...was:Mic positioning for choir (+Fletcher's HRTF session comments)

From: GuySonic (guysonic@aol.com) Newsgroups: rec.audio.pro Date: 1999/02/06

In article <36B8F703.7748@mercenary.com>, Fletcher <Fletcher@mercenary.com>
writes:

>Subject: Hail Curtis...was:Mic positioning for choir (+Fletcher's HRTF
>session comments)
>From: Fletcher <Fletcher@mercenary.com>
>Date: Wed, 03 Feb 1999 21:25:24 -0400
>
>>
>> In article <36b4ea6f.0@newsread3.dircon.co.uk>, "Frank Wood"
>> <woodf-l@dircon.co.uk> writes:
>>
>> >Subject: Re: Mic positioning for choir (+Fletcher's HRTF session comments)
>> >From: "Frank Wood" <woodf-l@dircon.co.uk>
>> >Date: Sun, 31 Jan 1999 23:41:10 -0000

>> >Spaced pairs can give you an awful hole in the middle. The classic way is a coincident pair of anything from straight cardioids to figure-of-eights, according to how much ambient (i.e. reflected) sound you want. Hypercardioids, even.
>> >

>> >I missed Fletcher's HRTF post, but as far as I can tell, this sounds like a way to add a directional character to omnis.
>

>I missed it too...Like I think I forgot to do it...so let me try and make amens. Leonard was kind enough to send me a dummy head with a couple of the mics that clip on where the ears would go on a human. We named the brother "Curtis".
>
>After properly dressing Curtis (do-rag, shades, mouth complete with cigarette-unlit), Curtis was led around the "Mainstage" recording space.

>This was setup number 2 of three, it was on the 100'x40' rehearsal stage at Longview Farm. Along with a rather excellent sounding set of Ludwig drums were 2-SVT's, 2 of my custom Aerosmith (started as HD-130's and got louder from there) amps...and a fairly decent EAW FOH system so you could hear the drums over all that racket. Average SPL for the stage >was about 122db.
>
>The SR sytem also had the loops (on the songs they were used...not all of 'um) coming out of it, as well as a reference vocal. The kit was mic'd with a Neumann M-147 about 4'(ish) in front of the kit, and a Neumann KM-54 over the drummers right shoulder.
>
>The PA was fed by a Shure 57 for the snare, a Shure 57 for the kick, a 421 for the rack tom, and a 421 for the floor tom. It took about a day to get the drums sounding really good out of the PA (yeah, 57's...what of it?), but when we got it sounding good...Curtis put you in the environment that was that stage, sans the pain of being up there for an extended period of time (my tinnitus was ringing nearly as loud as the damn stage by the end of 2 weeks).
>
>Last weekend past was the start of vocal and percussion overdubs, I was amazed at the feeling of space Curtis was able to capture. Curtis was even better after some rest (which is not always the case in my past experience). It quite captured the feeling of being there, added a thickness and dimension to the recording that I'm not too sure could have been captured by another method...it was more than just a clarity, it really gave me a sense of being there.
>
>Whatever the thing is called...Mr. LiteGuy, HRTF, DSM, Curtis...he rules. I think one of the things that amazed me most was that when I first put Curtis up, I did it "intellectually"...I thought about where I thought was going to sound best for Curtis...after two days of fucking with the sound and balance of the band on the stage, I noticed myself gravitating to one spot on the stage, and listening from there. I got a really good balance of drums, guitar and bass at this one spot...so when Curtis went to that spot, I got the same sound in the control room. The "intellectual" spot wasn't bad...but the balance was a bit off...more like "drum ambience" mics, which wasn't my intended purpose. The intention being to get a balance of the whole band playing...which I got, but I got a bit more.

>I've lightly messed with it a little bit, nothing major, but tried a little compression for the after session ruff mixes...they needed a little lower mids pulled after adding a bit of compression, but

that has more to do with the environment than Curtis. The rear KM-54 required a similar pull on some low mids as well. The M-147 just sat there like the cardioid lumox it is and did an excellent impression of a 47fet with too much output...in other words, it sounded really good (he said breaking an arm patting himself on the back...OK it sounded like I wanted it to sound...no, this will never get a Grammy nomination...but I think it sounded right for the band...enough).
>
>I don't know what Curtis sells for...but I do believe he's on the wish list for the future. I don't really think I want to cut more basics without him...he's become a real pal!!
>
>Sorry this took so long to post, but I think only Leonard was waiting for it, and frankly, I'm really glad to have had the opportunity to review the tracks again before posting this.
>--
>Fletcher
>Mercenary Audio
>TEL: 508-543-0069
>FAX: 508-543-9670
>http://www.mercenary.com

Well, you're right (as usual), it seems like I may be the only one waiting or cared for hearing about how an (HRTF) LiteGUY baffle with DSM mic actually worked within a real recording session.

I for one appreciate your kind effort and care in writing a report that also provided much insight to session and post session work in general that should've been of interest to others (besides you, me, and a maybe few 'silent' others) within this NG who have interest to learn about acoustically mic'd ambient stereo recording methods like using "Curtis" and spot mics for bringing a session within acceptable focus.

Of main importance is your being able to hear a good ambient mix at some postion and then get that exact same sound recorded using LiteGUY 'Curtis' in the same postion. The LiteGUY baffled spaced omni mic feature of being able to just listen normally (not having to plug one ear and then second guess or trial and error reposition the mic as Scott often suggests) for a good mic placement position should be of great value for recordings that need be quickly setup and allow little time for trial and error refinements. No other stereo mic method (other than a LiteGUY type mic) will 'always' and reliably give that "what you hear is what you actually get recorded" ability.

Fletcher, thank you for the informative post and for having the courage to try something new.

Best Regards in Sound & Music, Leonard Lombardo
Sonic Studios(tm) "Making Audio History With DSM(tm) Microphones"

Subject: Re: Hail Curtis...was:Mic positioning for choir (+Fletcher's HRTF session com...

From: hank alrich (walkinay@thegrid.net) Message 2 in thread Newsgroups: rec.audio.pro Date: 1999/02/06

GuySonic <guysonic@aol.com> wrote:

> Well, you're right (as usual), it seems like I may be the only one waiting or cared for hearing about how an (HRTF) LiteGUY baffle with DSM mic actually worked within a real recording session.

Wrong! <g> I was looking forward to Fletcher's reaction to your setup, and now it's archived.

Damn, so many mics; so little money!

--
hank - secret mountain
Note: the rec.audio.pro FAQ is at http://recordist.com/rap-faq/current
Read it and reap!

Subject: Re: Another $400 mike question (Pipe Organ Recording)

From: GuySonic (guysonic@aol.com) Newsgroups: rec.audio.pro Date: 1999/11/07

In article <19991105193922.27390.00001179@ng-fb1.aol.com>, nhsns@aol.com (Nhsns) writes:

>Subject: Another $400 mike question
>From: nhsns@aol.com (Nhsns)
>Date: 06 Nov 1999 00:39:22 GMT
>
>What is the best single-point stereo microphone less than $400. It must NOT require phantom power--or any strange, difficult to obtain battery.
>
>The best recommendation I've received so far is the Sony ECM-999. If you have an alternate choice, please let me know.
>
>Good bass performance is mandatory. This mike will be used for pipe organ recording.
>
>Thanks a bunch,
>
>Norm Strong (nhsns@aol.com) or (norm@scn.org)
>2528 31st South, Seattle WA 98l44
>

Your post for a reasonable way to record the full bandwidth of a pipe organ in stereo is best done with two full pressure type omni microphones.

None of the single point microphones, ribbon microphones, or any of the other types posted as suggestions will work out for at least lack of pressure type bass response regardless of what the posts claim and for other just as important stereo imaging requirements.

However, using these two (pressure type) omni microphones spaced out in some manner is not enough for recording the spatial ambient sound in stereo that's also very important for making a satisfying large size acoustic instrument recording.

A baffle needs be used placed between the two mics for the ambient stereo aspect to also be recorded faithfully.

Jecklin Disk type baffles are OK and far better than NO Baffling, but lack some important (HRTF) features for consistent results.

My site listed below has tips, reviews, mics, baffles, and sample sounds of pipe organ recorded with Sonic Studios (my own company) DSM designed mic systems.

Suggested mic model: DSM-6/H (headworn or using the LiteGUY HRTF Baffle)
Some URL's to view:
http://www.sonicstudios.com/mp3.htm (see St. James Cathedral, Seattle recording)

http://www.sonicstudios.com/liteguy.htm

http://www.sonicstudios.com/reviews.htm

Mic models suited to your music or sound recording tastes are listed at:http://www.sonicstudios.com/dsm.htm

Powering & bass filter considerations are discussed at: http://www.sonicstudios.com/pa_x.htm

E-mail me with questions and about your current recording deck/preamplifier equipment for best system fit suggestions.

> Best Regards in Sound & Music, Leonard Lombardo
> Sonic Studios(tm) "Making Audio History With DSM(tm) Microphones"
> Ph.541-459-8839 USA Free:1-888-875-4976 WEB: www.sonicstudios.com

Subject: Portable Audio....

From: **Jaspenn (jaspenn@aol.com)** Newsgroups: rec.audio.pro Date: 1999/09/19

> Please excuse me if this is not the proper NG, but I'm looking for suggestions and information on portable audio recording hardware and software. I will be recording audio (non-musical) in various places and will want to post edited versions to a web site on a routine, daily basis. I will need to record as much as 2 or 3 hours at a time (on tape?) and then send excerpts to a web site. Can this be done on a lap top? Or will a seperate recorder be more likely? Any advice and recommendations will be appreciated, especially by email. Thank you.

From: GuySonic (guysonic@aol.com)Subject: Re: Portable Audio....

I know that at least some would say this is the right place to post this subject.

Your desire for recording (stereo ambient sounds?) might be well served by using a portable MD like Sharp's MD-MS722 or (better for long duration & highest quality) Sony PCM-M1 DAT deck.

(PRODUCT PITCH) Many who are doing similar live field recordings are finding Sonic Studios (my own company & product) HRTF baffled or headworn DSM stereo mics a perfect fit

Sonic Studios' Site content is extensive and dedicated to this type of recording with reviews, recording tips, recommended hardware, and .MP3 sounds/Music done by others and myself (with taking my own advice on recording methods/mics/decks). Take a look there for some examples and good ideas.

See:
http://www.sonicstudios.com/mp3.htm

http://www.sonicstudios.com/reviews.htm

Recording directly to a Laptop is feasible *(maybe consider using an Opcode DATport or currently available www.M-Audio.com external USB soundcard device or similar on a laptop)*, but I'd strongly suggest recording to DAT tape while digitally outputting USB ported audio as described at: (http://www.sonicstudios.com/datport.htm). Laptop live audio recording seems to be not nearly reliable enough and too power hungry for casual portable use.

> The affordable and practical Sony PCM-M1 DAT deck is ideal for making reliable very high quality 2-track 2-3 hour length recordings with or without the laptop in the loop.

As a side note along with this topic, I did a recent on location (Jazz piano, drum, bass trio) session work using HRTF LiteGUY baffled DSM-6S/H mic on a boom, custom DC servo preamplifier, and a full size Sony R-500 DAT deck running tape but with digital output to a (all UPS powered!) Laptop DAW+CD-R system with a DATport digital I/O.

What I really liked about having the session (successfully!) already on a DAW Hard Drive was the ability to hand the client not one, but two 50+ minute length CD-Rs of the most promising takes within 45 minutes of the session's end! The customer (who was very time-limited as to get to his evening performance) took home the 2 CD-Rs (25 total tracks) for later careful listening and this allowed the next day "backup" sessions to be deemed unnecessary.

The quickly done rough-track CD-R allowed good customer assessment of usefulness of the session's recorded tracks and completion of the all goals in that first day. It took only about 4 more hours to edit/master a proper 60+ minute worth of CD-R tracks suitable for assembly and duplication without once
needing to transfer anything from the DAT "backup" tapes.

This is all very good when the Laptop recording system works, but just one little problem during the session with the Laptop, its software, or interface would likely have made this a two day session (like NO customer take-home CD-R & QA) with double or triple the time required to transfer material from the DAT "real-time" tape to the DAW edit/mastering process.

While some of you may have debugged and now use a reliable "fixed_in_place" studio system with this direct to hard drive edit advantage, mine was a on-location quick setup job that just happened to have everything work like it did at "rehearsal" back at the ranch. I just wouldn't bet everything on a laptop system working flawlessly every time; better to have a DAT tape copy also running just in case.

Best Regards in Sound & Music, Leonard Lombardo
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Subject: tuning near-fields

From: Christian Esteves (esteves32@worldnet.att.net) Newsgroups: rec.audio.pro Date: 1999/09/15

>newbie...can anyone inform me how to correctly tune near field monitors....(a brief explanation much appreciated)

This generally is not done.

Close monitors are left natural, as it is assumed that they will not be running so loudly that faults in the acoustics of the room will be overly abundant. (Though this is not always the case).

'Tuning', applies to the acoustic environment in which you have placed a speaker. That tuning is preferably done by modifying the physical shape of the room and it's reflectivity. IOW, you are tuning the room by making physical changes, not tuning the speaker itself.

Inserting an EQ in the path of the control room playback in order to make the final, minute adjustments when using mid field or distant (large) monitors, may be what you are referring to. It's the 'fine-tuning' that might still need to come after the physical (structural) changes and surface treatments are made. It's not recommended and should be considered last, after other methods of taming the room are exhausted.

After sentence one, that's about as brief as it gets.

David Morgan (MAMS)
Morgan Audio Media Service
Dallas, TX (972) 622-1972

____
Message 3 in thread
From: Bill Roberts (wroberts@grove.ufl.edu) Subject: Re: tuning near-fields Newsgroups: rec.audio.pro Date: 1999/09/15

Christian Esteves wrote:
>
> newbie...can anyone inform me how to correctly tune near field monitors....(a brief explanation much appreciated)

I assume you mean you have monitors that have switches offering options of increased highs, etc.

With regard to bass, this is selected according to whether you are placing them in a corner (you need the most bass cut to compensate for this), against a wall (some bass cut), or in free air.

With regard to treble, I would leave it flat and then see how the reproduced sound (from a recording that is very natural sounding, perhaps one you made yourself using a very neutral mic, or a CD) compares to natural. If the sound from the speakers has more or less treble or bass than the original sound then change the settings.

Also see how mixes seem to translate when you play them in cars, home stereos, etc. If mixes played outside the studio don't have enough highs, for example, though they sounded right in the studio, then the monitors are adding highs you don't want and you should change the setting.

Problem is if you are not experienced with how things "should" sound then you really cannot do this. If you make the studio monitors sound like a home stereo that has the graphic EQ set in a smiley face (boosted top and bottom and scooped in the middle) and with the loudness button on, you will be screwed. And if that is your idea of how things should sound, you will be screwed until you learn otherwise.

-- Bill

Message 4 in thread
From: Randy (rkirk@rocketmail.com) Subject: Re: tuning near-fields Newsgroups: rec.audio.pro Date: 1999/09/16

Bill Roberts <wroberts@grove.ufl.edu> wrote in message
news:37E050A6.C2261ADC@grove.ufl.edu...

If you make the studio monitors sound like a home stereo that has the graphic EQ
> set in a smiley face (boosted top and bottom and scooped in the middle) and with the loudness
> button on, you will be screwed. And if that is your idea of how things should sound, you will
> be screwed until you learn otherwise.

hmmm. I'm never been a full-time engineer, but I've recorded and mixed a lot of demos over the years and I don't quite agree. While I do mix with my near field moniters and amp EQ'd flat, at times I have difficulty getting the right bass level that sounds great on other stereos. With small moniters, the bass response (obviously) isn't there.

So.. (purists please disregard) in addition to monitering flat, I also moniter with the loudness switch ON (at lower volumes), if for no other reason to provide a different POINT OF REFERENCE, not unlike playing the mix on a different system. Since many people (myself included) tend to boost bass when listening for pleasure (especially electronic music), why not also moniter with same settings as you'd normally listen to, and are most accustomed to? My point here is that boosted bass has become it's own reference standard for the general public, so why not use it as a tool?

I've found that the easiest way to determine whether or not I've got too many lows in the mix is to moniter with the loudness switch ON. If I've mixed the bass too loud while monitering flat, the loudness button blows it WAY out of proportion -- an obvious sign that I should ease back on the lows. The bottom line is to have points of reference that you can work with.

By the way, I probably wouldn't do this if I mixed with a subwoofer and/or larger moniters.

OK bring on the flames, tinnitus-heads. ;)

Message 5 in thread

From: GuySonic (guysonic@aol.com) Subject: Re: tuning near-fields Newsgroups: rec.audio.pro Date: 1999/09/16

Applying a MONO FM interstation sound (or pink noise) to each speaker after placement is fairly certain will help setup the stereo imaging aspect.

My technique is to then listen 'dead-center' at the listening position for how this mono signal images. I find it more ideal when the noise appears to come from the exact center and seems about 1/4 to 1/2 the speaker spacing width at the optimum listening position.

The monitors are angle adjusted (towed inward) to make the mono "pink noise" sound as narrow or as wide as you desire and this angle will vary with the speaker and the frequency; pink noise has virtually all frequencies and serves to test the full range frequency imaging aspect of any speaker and the speaker position.

I personally like fairly wide space speakers that're usually at least as wide spaced as distant from my listening position and angled to focus 'just in front' of my head.

Usually, the closer you're positioned to the monitors, the more critical the speaker/listening positioning for accurate imaging. The larger the monitor speaker, the more distance is required to widen the usable listening position. Getting at least a full 12 inches of critical listener position side-to-side width might be considered quite adequate for nearfield monitor setups. *(also see surround speaker positioning suggestions on the tips page)*

Best Regards in Sound & Music, Leonard Lombardo
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Subject: Re: Question from a new poster Message 9 in thread

From: GuySonic (guysonic@aol.com) Newsgroups: rec.audio.pro Date: 1999/09/10

In article <19990909193310.25386.00006342@ng-fz1.aol.com>, hatnyc62@aol.com (Hat NYC 62) writes:

>Subject: Question from a new poster
>From: hatnyc62@aol.com (Hat NYC 62)
>Date: 09 Sep 1999 23:33:10 GMT
>
>Do you know what the big labels like Sony are doing when they record clarinet with piano or with orchestra to 'sweeten' the sound of the performer. I am mostly speaking in terms of microphone placement. If you have heard some of the better known performers live and on records, you can hear that the engineers have found a flattering way to record some of the clarinetists today.
>
>I am asking because a friend of mine and I have been doing some experimental recordings with some excellent equipment (neumann u89, millenia media pre, apogee 24 bit converter). The sound we get is mostly accurate, but rather clinical, definitely not 'flattering' to the clarinet sound. I was wondering if there was something we hadn't thought of.
>
>Thanks in advance for your thoughts.
>
>

You might consider a completely different microphone method that records an ambient stereo more nearly to how you hear the sound at the recording position.

This type of recording (download some of the .mp3 music samples on my site) consistently provide a very complimentary sound of all acoustic instruments at fairly close to further out distances.

Distance or mic position is a variable that is best determined by actually normally listening for the better mic positions. The mics and methods so far discussed will not allow you this convenience as they do not record like or what you're hearing.

A mix of direct instrument to acceptable ambient sounds will be different for each room, instrument, and desired effects appropriate for the composition. Listen for what's acceptable, then record it at this chosen position with this stereo microphone.

DSM-6S/EH or /H models are suggested either headworn or with the HRTF GUY or LiteGUY mounting baffle.

Best Regards in Sound & Music, Leonard Lombardo
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Message 3 in thread

Subject: Re: Recording a small choir & instruments

From: Scott Dorsey (kludge@netcom.com) Newsgroups: rec.audio.proDate: 1999/04/11

In article <1dq3sll.4gqbfh13pd0k2N@ppp00664.01019freenet.de> saetc@gmx.de (Hannes) writes:

> \>
> \>I am a student in audio technology in Germany. Please excuse my simple English, my last lesson took place really long ago.

My German is far worse. You are doing much better than I would be able to do in German.

> \>I am going to do a recording in a church. There will be: - a choir, consisting of 5 girls - two instruments: an acoustic guitar and a small drum, both played by members of the choir.
> \>
> \>The problem is that the girls are not used in working with headphones. I fear that they will not be able to perform properly when I first record the instruments and then record the choir hearing the instruments and themselves on the headphones.

Why do you want to do this in multiple takes with overdubs? Why do you want to use headphones anyway?

This seems like a very appropriate use for a single microphone pair, just recording the whole ensemble as one. You have musicians who are used to playing with one another and who naturally balance themselves, so don't wreck up the balances. Just set up an ORTF pair and position it until you get a good mix.
--scott
--
"C'est un Nagra. C'est suisse, et tres, tres precis."

Message 4 in thread

Subject: Re: Recording a small choir & instruments

From: Lars Kr. Tofastrud (romakust@online.no) Newsgroups: rec.audio.pro Date: 1999/04/11

Mr Dorsey is absolutely right!

No reason at all to use headphones and stuff

Use a pair of Earthworks QTC1 and a good mic pre-amp and a nice recorder. (Tascam 24bit DAT or some PC based stuff with at least 20 bit resolution)

you should get a natural and correct balance if they are capable of playing/singing together (I suppose they do)

Stereo will probably blow you away! Listen to your recording in a nice mastering quality hi-fi room and you will be thrilled!

Best regards
Lars Tofastrud

Message 5 in thread

Subject: Re: Recording a small choir & instruments

From: GuySonic (guysonic@aol.com) Newsgroups: rec.audio.pro Date: 1999/04/11

While I agree with Scott and Lars on the stereo pair approach as best, the ORTF pair is directional and excludes side/rear ambiance with also inducing undesirable and quite audible phase distortions. Setup is a bitch as you never really know what you're going to get and it sounds different with each playback system. Trusting what you hear during setup is not going to provide much security with this microphone method.

Just spacing a single pair matched omni removes the phase distortions problem nicely, but this mic method presents frequency dependent phase cancellations (due to the spacing) that are audible to very audible with great difficulty or impossible to remove after the fact with any satisfaction.

Setup is a bit easier with spaced omni method as the flanging is more consistently audible allowing the adjustments to be made with much necessary rehearsal as the sound stage+ambient is active in producing an acceptable distance from the stage and spacing of the mic array. But, setup is still a bitch as you can still never be completely sure of he final product even with much setup time. Trusting what you hear is not part of this method of mic technique.

With both these previous discussed methods, much luck and experience is needed to get OK results.

An advanced method of mic technique again uses two matched omni, but with a HRTF baffle between; and NO, this is not a disk or sphere baffle. This HRTF baffle eliminates audible flanging common with spaced omni and makes setup a snap as just listening with both ears in a normal "surround-sound" mode to what the sound is like at any chosen mic position will reflect exactly the recorded sound as reproduced by a wide variety of speakers and headphones. Setup is most easy and reliable because the mics record in a manner that replicates how sound is heard, but is not a binaural method with binaural limitations of playback. DSM recorded Sound is naturally Pro Logic encoded for playback with full 360 degree ambience available.

While my company provides both mics and HRTF baffle hardware specialized for this purpose. Your choice of paired omni mic is your own, but HRTF baffles of proper design are only available from Sonic Studios or use your own head to baffle the similar mics.

A discussion of the HRTF recording method is found at:
http://www.sonicstudios.com/multitrk.htm

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Subject: Re: Recreating Spatiality in Mix View

From: GuySonic (guysonic@aol.com) Newsgroups: rec.audio.proDate: 1999/04/11

In article <cclee-1004992302470001@a27-44.itis.com>, cclee@itis.com (Chia Chin Lee) writes:

>Subject: Recreating Spatiality in Mix
>From: cclee@itis.com (Chia Chin Lee)
>Date: 11 Apr 1999 04:03:40 GMT
>
>I'm experimenting with ways to achieve better spatiality in my mixes, and I thought I'd turn to you guys for some thoughts and advice.
>
>First, some background concerning what I'm aiming for. I'm working with electronically-realized orchestral music. After a/b-ing my music with what I consider to be good mixes, I have found one *major* difference.
>Although my mixes have good balance in terms of Left-Right relationships, the mix is severely lacking in a sense of Front and Back.
>
>Granted, most of the music I'm comparing against are performed live, in a real acoustical space, so I'm not really being fair. What I'm aiming to do is to simply give my sample-based music a better sense of spatial depth
>through panning and reverb.
>
>Here's what I'm doing right now:
>I've attempted to create a sense of space by submixing 5 sections (Front, Near

Front, Middle, Near Back, Back) of the orchestra. All the individual instruments are panned according to where they sit in the orchestra, and >placed in each section of the submix. I am mixing the Front section (First Violin, Cello, Bass) with the least reverb, with each submix increasing in reverb, until the Back (mostly Percussions) which receives the most reverb treatment.
>
>As I said, I am quite happy with the Left-Right result of the mix (panning is easy... :), but the Front-Back is sounding less distinct than I'd like. Ideally, I'd like to be able to close my eyes and have a mental image of the placement of the source of the sound.
>
>
>If you have any ideas on how to recreate better spatiality in a mix, I'd appreciate it.
>
>Thank you!
>

While full Dolby 5.1 processor encoding remains an option for doing something like this, my site has an article on how to do this in a reasonably good sounding ambient by staging playback speakers of the tracks you want to
localize in real dimensional space. This method is prime for taking electronically synth sounds with no dimensional acoustics and adding the 3rd dimension via re-recording with a 3-D stereo microphone method that then provides the final mix with all the needed realism of hearing this as an acoustic event at the 3-D stereo mic position.

The psycho-acoustical cues of dimensional hearing (like what we use to localize sounds in real acoustic space) is what the DSM microphone records using a specialized (patented) Head Related Transfer Function (HRTF) baffling of miniature spaced omni (the DSM mics).

URL for this article is: http://www.sonicstudios.com/multitrk.htm

The hardware to do this type of recording is also featured on the site listed below.

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Subject: Re: Recording a Piano/Vocal Combo

From: GuySonic (guysonic@aol.com) Newsgroups: rec.audio.pro Date: 1999/03/31

In article <3700EB3F.E998F9A@millerthomson.ca>, Jordan Slator <jazz@v-wave.com writes:

>Subject: Recording a Piano/Vocal Combo
>From: Jordan Slator <jazz@v-wave.com>
>Date: Tue, 30 Mar 1999 08:18:23 -0700
>
>I've been asked to record a few songs by a friend's piano/vocal duo,
>and I need some advice on both equipment and technique. My own
>equipment (Makie, SM57s, etc.) is basically geared for budget
>basement-rock demo type stuff, but I want something better quality for
>this recording. They want to press a few CD's and use them for
>demo/portfolio purposes.
>

I'd approach this in an easy and natural way using one or both the following suggestions.

Have the vocalist wear a DSM HRTF stereo microphone and stand at a position where the piano (facing the open side) is considered a good mix with the vocal (listen on phones to the stereo to determine this position). This method is beneficial with vocalists of all kinds of ability and you'll just need to listen to the ambient piano mix with the vocal to determine where the vocalist is heard at a good level the piano; consider this THE MIX (all acoustic here) with hearing a good balance of the two. Very simple and fairly foolproof.

Second method is to place the vocalist in front of the open side of the piano facing away towards the DSM headworn by you, an assistant, or baffle worn on the LiteGUY HRTF baffle. Wearing this stereo mic yourself will instantly give you the 'as-you-heard-it' recorded mix (no headphone monitoring necessary, but using sealed earbuds is a possibility for the insecure when self wearing this mic). Spacing the vocalist's distance from the piano adjusts the vocal to piano mix, and spacing the DSM microphone from the vocalist will give the appropriate vocal+piano to ambient mix for an excellent and natural acoustic stereo recording that will hold up consistently as a good sound under many different playback conditions. A vocalist with good projection and who knows how to use the room ambient is a plus with this second method

Details about the DSM microphone and HRTF stereo recording method are available on my site or questions answered directly by phone or E-mail.

There are plenty of ways to record the sounds of piano and vocal, but you'll need plenty of setup time, rehearsal time, more equipment, and a generous amount of luck to succeed as well with other approaches that don't record in a natural as-you-hear-sound manner.

It's always your choice (or should be if acting professional about this), but being open to suggestions is the first step in knowing your options and learning about a very rewarding, but often way too (unnecessarily so) convoluted technical subject.

The ability to recognize and know the best solutions in this field of interest will make achieving recording satisfaction far easier and more consistent; allowing you to succeed under a very wide range of situations. The ability to hear just what a recording method is doing with just your normal hearing (no double thinking or impairing of the hearing to replicate some odd mic response or pattern, or needing special monitoring systems, or having to muffle an acoustic instrument), the more quickly satisfied you'll be with recording acoustic sounds.

Now for all the alternatives to consider........ Now gentleman, start (or continue) your replies....

Regards in Sound & Music Recording,
Leonard Lombardo

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Subject: Re: Help Recording Vocals w/Reverb

From: GuySonic (guysonic@aol.com) Newsgroups: rec.audio.pro Date: 1999/03/15

In article <kludgeF8M7H5.GIJ@netcom.com>, kludge@netcom.com (Scott Dorsey) writes:

>Subject: Re: Help Recording Vocals w/Reverb
>From: kludge@netcom.com (Scott Dorsey)
>Date: Mon, 15 Mar 1999 02:41:29 GMT

>
>In article <36ec6918.4919901@news.erols.com> markcts@erols.com (Mark) writes:
>>I'm trying to lay down a vocal track with reverb (Alesis Wedge).
>>During the actual recording the monitor vocals sound crisp, deep and
>>solid, but upon playback from my analog MT100II 4-track they seem kind
>>of washed out, with a heavy low end (yes, dbx and EQ are out). This

>>happens no matter what Wedge patches I use.
>

>**Try recording the vocals dry.** Then, on playback listen to them. Are they okay? If they are clean but dry, then try using the Wedge on mixdown and see what it sounds like.
>
>If it still doesn't sound right, you may just be a victim of poor quality digital reverb. But first get it to sound right before the reverb.
>
>And, of course, realize that your voice always sound thinner on tape than it does in your head....
>--scott
>--
>"C'est un Nagra. C'est suisse, et tres, tres precis."
>

It's interesting you mentioned this as there is a way to record your voice exactly as it sounds to yourself.

Danny Glover (the actor/producer; Lethal Weapon and many others) is doing just that in LA (this week) as directed by the guys at Skywalker Sound using a set of DSM microphones worn by Danny as he narrates in a sound booth. This gives an excellent you-are-inside the narrator's head dry stereo soundtrack that can
be enhanced later with EFX to sound like it's taking place in different ambiences (to fit the visuals).

Vocalists can also use the DSM headworn technique and shifting the microphones forward (nearer the temples) or backward gives a tonal shift that's brighter near the temple positioning.

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Subject: Re: Orchestra micing question

From: GuySonic (guysonic@aol.com) Newsgroups: rec.audio.pro Date: 1999/02/12

In article <kludgeF6yxM7.LJL@netcom.com>, kludge@netcom.com (Scott Dorsey) writes:

>subject: Re: Orchestra micing question
>From: kludge@netcom.com (Scott Dorsey)
>Date: Thu, 11 Feb 1999 02:30:06 GMT
>

>In article <36C22D75.521D8019@indigo.ie> Daire Winston

<beech@indigo.ie writes:
>>I'm going to record a full orchestra, rhythm section and lead vocalist
>>in a live in a concert hall this summer and I would appreciate any
>>thoughts as to how to mic the individual sections within the orch. I
>>will obviously use a stereo pair of room mics but what it we need to
>>boost the 1st violins or the flutes etc. at the mix.

>
>Then, you tell the conductor, "Hey, boost the violins" and he signals them to play louder.
>
>You get the orchestra pickup balanced, and he'll get the orchestra itself balanced. That's his job.
>
>Spot-miking is possible, and depending on the vocalist and the rhythm section (and whether they get PA), you may well have to spot mike those. But the orchestra itself is one entity, and you should mike it like one entity.
>
>There are good reasons to spot-mike orchestral sections, mostly having to do with bad hall acoustics or PA leakage, but it's not for the faint of heart, and I don't recommend doing it on your first orchestral gig.
>It's very easy to screw up balances and it's impossible not to screw up tonality.
>--scott
>
>--
>"C'est un Nagra. C'est suisse, et tres, tres precis."
>
>

Recording a complex ensemble of mostly acoustic instruments in a satisfying manner has always been extra challenging. A different philosophy of microphone for recording this in pure 2-channel stereo has many advantages and some disadvantages.

Orchestra, chorus, solo piano types of acoustic recording is most critically judged against any live memory impressions we chance to have at least once if not many times in a lifetime. Most other popular forms of recorded music have little need for any 'reality check' with needing few 'ambient acoustic' qualities to be satisfying. As such, popular using of close mic'd and multitrack mix + EFX in post processing provides an 'electronic' but satisfying non-reality quality that is quite acceptable.

As Scott more than hinted in his reply, using spot mics to add volume to into a predominately natural acoustic mix, easily throws at least the 'tonal qualities' into a spin that is rarely but partially recoverable.

Using a single stereo microphone positioned 'as well as possible' to record all sounds with no other microphone(s) in the mix gives the most consistent satisfaction under less than optimum orchestra staging or 'allowed' mic position during performance. It's far less disturbing to notice one of the sections are a bit too loud or soft than to have to endure an entire recording of off color tones and displaced images.

I would suggest recording any and all practice sessions, especially those in the performance hall to fine tune what works best in simple and natural sounding stereo without spot mics. If the vocals are PA mic'd, consider recording a mono track of this (and only the vocal) to be worked carefully into the stereo tracks in post or if feeling lucky, mix it into the stereo during the recording.

My site describes 'my own favorite' (ambient) stereo microphone for this type of recording. This is a paired HRTF baffled omni method that's unique enough to be patented. The LiteGUY mic baffle (the dual omni mic mount) is black Ultra-Suede and can 'disappear' if not in direct line-of-sight of the camera, allowing both satisfying sound and easy placement that's far less critical than most other stereo mic configurations you could also consider using.

In my opinion, it's far better to go for the more satisfying live feel of naturally mic'd stereo than to go for getting everything perfect in volume and losing most of the 'musical emotion' content as mostly happens with orchestra spot mic mixed recordings. It's definitely a matter of choice of what's more important to the enjoyment of the music for the masses. Does this choice always have to be the recording engineer's notorious addiction to electronically equalizing recorded volume levels of orchestral elements?
This almost always excludes the more important (in my opinion) benefits and accesses able enjoyment of live 2- rack stereo recording that's more identical to our live experiences of this music.

I say, stand back a bit and record it in stereo like it sounds to most of us as if we had the best possible seat in the house! Ignore the need for perfect volume level of all elements and gain the ability of convincingly recording the 'experience'.

Best Regards in Sound & Music, Leonard Lombardo
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Subject: Re: mic cable quality question

From: GuySonic (guysonic@aol.com) Newsgroups: rec.audio.proDate: 1999/02/08

In article <79l67m$n7k$1@camel15.mindspring.com>, "morrison" ultrevex@mindspring.com writes:

>Subject: mic cable quality question
>From: "morrison" <ultrevex@mindspring.com>
>Date: Sun, 7 Feb 1999 18:09:12 -0500
>
>Yesterday, I was tracking female vox. The singer was in my small booth, and
>signal path was mic(soundelux U195)-25 foot no-name cable-sytek mic pre- a/d
>converter-DAW. This combo has only been up and running for several days and
>I've been getting consistently better results with time(as expected). Later
>in the session, we were fleshing out an idea and I moved her into the
>control room. I left the long cable in the booth and grabbed a 12 foot
>cable. equally 'no-name', non esoteric, unimpressive local mom and pop
>music store type mic cable. The sound difference was incredible. The
>shorter cable was letting much more presence through, much more signal, just
>much more everything. My questions: barring the fact that the first cable
>is shot, what was I hearing? Capacitance from the long cable? Crummy cable
>quality? I plan on going thru my box of xlr's and A/B'ing the lot but does
>anyone have any wisdom to impart in the meantime? The first cable didn't
>sound bad until I heard the shorter one!
>
>Also, I doubt I'll replace every XLR cable in the studio but I imagine it
>would be a good idea to get one 'snob' cable for overdubs. Like most small
>places, once basic tracks are cut it's one overdub at a time through one
>mic(at a time), one pre(at a time)and now, I guess, one cable?
>
>How about cable favorites. I dont roll my own and dont plan on dropping a
>stupid amount of money on a single cable. anyone heard any appreciable
>differences?
>
>thanks
>Kevin Morrison
>
>

Even with identical cable type, shorter is usually better in general for having less high frequency attenuation and the audible effects of cable reflections is less noticeable with shorter lengths. Some cable types are better than others of course, using less of any type is still preferred over lack of benefits found using cable length to any known advantage other than getting from here to there.

You can minimize mic cable effects by using a short cable to a mic preamplifier that in turn drives whatever long cable is found necessary to access the mixer or recorder. Preamplifiers, operating at driving cable at line levels, have more cable drive/dampening ability than most microphone outputs. This consistently produces a cleaner, more detailed sounding 'mic feed' with most any quality of cable type.

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Subject: Re: mic cable quality question

From: GuySonic (guysonic@aol.com) Newsgroups: rec.audio.proDate: 1999/02/08

In article <79m3v1$170@bgtnsc03.worldnet.att.net>, "J. Russell Lemon" lemon.j.russell@Worldnet.ATT.net writes:

>Subject: Re: mic cable quality question
>From: "J. Russell Lemon" <lemon.j.russell@Worldnet.ATT.net>
>Date: 8 Feb 1999 07:32:49 GMT
>
>I probably don't know what I am taking about, but I would suggest getting a cable that can pass the AES digital format. Such cables have lower capacitance then many old audio cables, and a more uniform impedance. A cable that can pass AES digital without errors should not lose presence when used for audio. Be sure to terminate in image impedance.
>
> --- Russ
>
>

Suitable AES cable is typically precision and 100 ohms impedance. Driving & Terminating all cables at their nominal impedance is good engineering but mpractical for most electronics rated for 600 ohms or higher driving ability.

This has always been the problem with effects of audio due to cable effects, they're rarely terminated properly or can be terminated properly by typical microphones and preamplifiers. As a result, signals bounce and ring causing all sorts of interesting effects, but this state of affairs is has dubious value on the quality of all audio subjected to even moderate cable lengths.

I at one time designed some very different audio/video cable that didn't require termination, but did require good capacitance driving ability from the source; 600

ohm (or lower) driving ability usually worked well. Beldon may consider this for manufacture sometime this year if all goes well.

M & K sound (many, many years ago) used 50 or 75 ohm RF/video cable driven by custom cable drivers and terminated at the cable impedance for remote location and studio work. This allowed them to run cable for many hundreds of feet to recording systems without any audible signal distortions or fear of electrical noise interference.

While the RF/Video cable itself solves many problems with signal quality and noise issues, finding excellent quality audio cable drivers for this task remains the real challenge.

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Subject: Re: How to stereo mic Grand Piano?

From: GuySonic (guysonic@aol.com) Newsgroups: rec.audio.pro Date: 1999/02/06
<< Subject: Re: How to stereo mic Grand Piano?

Eleven Shadows wrote:

> I think the ATM 31s are small diaphragm condensers that are similar to ATM 33Rs
> and ATM 33a mics, but someone correct me if I'm wrong. If that is the case, you
> can record the Yamaha with the lid open. There are a zillion ways to record a
> piano, but what I would do is try x-y over the piano, which usually works. I
> don't know what kind of music you are doing, but as a general rule of thumb, for
> more New Agey sort of stuff, get them farther away from the hammers; for certain
> kinds of classical and rock, get 'em a little closer. However, with a lot of
> classical, having those mics backed off and to the right of the piano player
> gives a more natural sound. Use your ear. A lot of people do spaced pairs,
> choosing to hover the mics over the strings that give it a more dynamic stereo
> spread.
>
> What I personally like doing is opening the lid, but having the mics backed off
> so that they are outside, and placing the mics at least three feet from the
> instrument. To my ear, this allows the sound to blend and sound more natural.
> If you have a really great sounding room, try a spaced pair of omnis. If you
> don't have access to that, you can try the two Audio Technicas. I like to try
> both spaced and x-y and see which each result gets.
>
> If it's rock piano, it's often heavily compressed to blend in with the rest of

> the music. In either case, the Audio Technicas are probably pretty bright mics,
> and I personally find that bright mics frequently help with the piano sound
> quite a bit.
> --
> Ken/Eleven Shadows, looking for a Super 8 camera and Super 8 projector
> ~~~~~~
> Eleven Shadows * ES songs on Real Audio * Music Reviews * Travels:
> Peru-Ladakh-Kashmir-India-HK * Tibet * Real Audio Radio Shows
> http://www.theeleventhhour.com/elevenshadows
> ~~~~~~
>>

As Ken stated "there's a zillion ways to mic a piano" for recording purposes. But there are just a few less than a 'zillion' ways to get an acceptable 'piano sound'. There are even fewer ways to mic for 'THE sound of a piano'.

Placing mics using the 'here and there' way, will reliable get that 'piano sound' we've grown accustomed to hearing in popular music tracks and there's almost a 'zillion' ways to approach this to get a unique 'piano sound' to fit the any occasion.

However, If wanting to mic for a realistic 'sound of a piano', the 'here and there' way of mic placement is NOT the way to go, but ANY of the stereo microphone methods discussed so far IS A WAY for intention of recording a convincing 'real piano' sound. To succeed with this however, presents far fewer ways (choices) of mics and 'using method' and presents a far greater 'skill level' challenge and/or being 'very lucky' to consistently get satisfactory results. Far from being impossible, it's at least much more
difficult for a number of good reasons.

The intention of recording piano realistically involves a stereo mic method as the output of these is two channels much like our own two channel hearing way of hearing sounds. So stereo mics need to record two different perspective that will most satisfy our natural hearing sense. For us to be convinced of hearing a real piano within a recording, the stereo mic must record sound in a 'unique' way that includes 'psycho-acoustical' information within the two tracks of recorded audio. While the 'psycho-acoustical' information necessary for us to hear a convincing sound of a piano is 'unique', the uniqueness of the stereo mic/method of using such, should not be TOO uniquely different from our own way of 'uniquely' hearing sounds.

And here's the rub of the stereo mic methods discussed so far: They can easily be far TOO unique and present only a 'partial set' of proper psycho-acoustical cues; often including (free of charge) a whole new unique set of strange (to our normal hearing) sound cues that are not 'coherent' or recognized (without doing 'mental' conversion type interpretation work) as part of a real sounding piano.

The stereo mics discussed so far are unique to each other (including us) in larger or smaller degree with mic placement rather critical to each new ambient situation. Because critical placement is often different with each 'type' of stereo mic/method (assuming the same ambient working condition), being able to listen yourself for an acceptable 'heard acoustic mix' of instrument and ambient (room, hall, etc.) is ALL IMPORTANT.

However, because of the degree of 'TOO much uniqueness' of each stereo method discussed, just listening will not reliably work unless you're (as mentioned earlier) very lucky. What you hear is NOT OFTEN ENOUGH what you'll record with stereo microphones and you'll need a lot of experience, luck, and/or time for the 'trial and error' record/playback procedure necessary to avoid disappointment from having assumed too much.

I would be much nicer to learn to quickly hear a microphone position (music + ambient mix), plunk the microphone right there, and roll tape (or spin hard drive) and be much more assured of getting what you heard because the stereo microphone is not so unique to our own perceptions of sound.

There's only one stereo microphone 'way' that'll consistently allow the 'what you hear is what you record' assumption regardless of situation. That microphone is a 'Head Related Transfer Function' (HRTF) type of stereo microphone that uses a unique baffle between two very small, precision matched omni mics.

This type of stereo microphone is rarely discussed or mentioned (at least here) perhaps because it's TOO much a 'no brain'R'??

Not being challenging, needing much skill, being lucky, or having the immense joy of doing multiple tests/retakes makes this type of stereo microphone hard to act expert about for sure, and may as such, be generally ignored by the standard knowledge base of available microphone experts.

As far as I can tell from being around here for over 5 years, it might just be working too well(!) dampening the joy of endless discussion of all the challenging ways 'uniqueness' in microphone 'perception' adds to our pursuit for convincingly real (ambient acoustic) recordings (if that's your aim). If mic/method solves a lot of previous problems, what will the 'problem solvers' now do? This remains a real 'bureaucratic type' challenge and seems worth much discussion of what to do next when 'favorite' discussed problems are threatened to be solved for good, making other options less accessible in appearing like good advice.

Fortunately, I'm here and again helping those who truly desire to make their recording more consistently real acoustically sounding with the experience, good advice, and the hardware to back up where my mouth is. As with many expert recordists, microphone companies refuse to adopt new microphone designs while the inventor still lives and breathes. If you doubt this, look for finding the persons responsible for the classic stereo microphone methods discussed here, (Blumlien, Soundfield, etc.) they've virtually all died years before any of these 'now highly discussed' methods were allowed real commercial production/availability or regarded as a mic technique worthy of discussion.

Things being as they are, no need to wait till I'm 'dead and gone' to get THE stereo microphone right now (only lacking any discussion of such from those 'teaching' the old standards of recording art), as I'm one of those very rare inventors that is able to produce products without the 'recording industry acceptance' due or scheduled sometime after my passing. My web site has the necessary details on THE stereo microphone that is very NON-assuming or in most ways non-unique to how we hear sounds; what your hear is exactly what you'll record; 'relearning' to trust normal hearing IS going to be tough on the 'old timers' used to sticking one finger in an ear between retakes. But while there's still life, learning is possible!

Please, don't all thank me at once for my dedicated efforts!

Just go out there and make it sound more real for the old GUY!!!

Best Regards in Sound & Music, Leonard Lombardo
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Subject: Re: How to stereo mic Grand Piano?

From: GuySonic (guysonic@aol.com) Newsgroups: rec.audio.proDate: 1999/02/08

In article <36BC08EC.86AD1A53@santa-monica-ca.com>, Eleven Shadows
elevenshadows@santa-monica-ca.com> writes:

>Subject: Re: How to stereo mic Grand Piano?
>From: Eleven Shadows <elevenshadows@santa-monica-ca.com>
>Date: Sat, 06 Feb 1999 01:18:38 -0800
>
>> This type of stereo microphone is rarely discussed or mentioned (at least here)
>> perhaps because it's TOO much a 'no brain'R'??
>
>The question was "how do I use ***the mics I have*** to get the best sound."
>This is the original question:
>
>> I have SM57,
>>C1000, two ATM31s, an old small D Audio technica omni, and an Astatic mic that
>>is a lot like a SM57. Going through a couple single channel Tube MPs into an
>>Akia DSP12 Hard disk recorder. Any suggestions on mic choice (other than get
>>better ones) and set up?
>
>Read the last sentence again.
>Great. Now read one more time and try and understand what the guy is saying.
>
>> Fortunately, I'm here and again helping those who truly desire to make their
>> recording more consistently real acoustically sounding with the experience,
>> good advice, and the hardware to back up where my mouth is.
>>
>> My web site has the necessary details on THE stereo microphone that is very
>> NON-assuming or in most ways non-unique to how we hear sounds; what your
hear
>> is exactly what you'll record; 'relearning' to trust normal hearing IS going to
>> be tough on the 'old timers' used to sticking one finger in an ear between
>> retakes. But while there's still life, learning is possible!
>>
>> Please, don't all thank me at once for my dedicated efforts!
>
>If you're that hot of an engineer, you should be able to get a good sound
>with inexpensive mics. Put your money where your mouth is and tell this guy how
>to get a good sound with the equipment that he has. I've tried to help him. Why
don't you?
>
>--
>Ken/Eleven Shadows, looking for a Super 8 camera and Super 8 projector
>~~~~~~
>Eleven Shadows * ES songs on Real Audio * Music Reviews * Travels:
>Peru-Ladakh-Kashmir-India-HK * Tibet * Real Audio Radio Shows

>http://www.theeleventhhour.com/elevenshadows
>~~~~~~
>

Ken, you're correct in pointing out not regarding his equipment limiting clauses enough and I was going mostly on the thread discussion that followed + the original post's stated desire to record a "piano in stereo". I hope the possible (over) dramatic portions of my past reply didn't completely mask the useful concepts presented.

> If you have a really great sounding room, try a spaced pair of omnis. If you
> don't have access to that, you can try the two Audio Technicas. I like to try
> both spaced and x-y and see which each result gets.
> ------------ (Ken's post)

>A piano is a mono source? I mic grands from 10-15 feet away using a
Blumlein array and even at that distance, the piano isn't mono. The
room
reverb isn't mono either but that is another discussion.

With piano, there can be a lot of experimentation. the sound can be a
distant room sound all the way up to the very close "head in the lid
sound" and this is just from a Blumlein array.

Other techniques such as miking above the pianist's head or from the
far
end of the piano will produce results that may be pleasing for different
styles of music.

AB, XY, MS, ORTF, Blumlein, and other techniques will give different
results. Some good some not-so-good.
-
Richard Kuschel
>

If he does have a pair of resonably matched omni (at least two of the same model/production run), then placing these two mics on a HRTF (like the LiteGUY) baffle will allow an easier way to record 'the sound of a piano" with minimum expense and skill level. If he doesn't have or want to get those two omni's you mentioned, then 'he can't get there from here' using any other type of mic if recording a realistic sounding piano is the goal with the available mic limitation imposed.

Then Richard opened the 'playing field' with naming (most all other) stereo microphone methods regardless of mic closet limitations. So here we are again. Ambient stereo recording of convincing quality involves correctly matching the perspectives of the microphone method with our own as accurately as is possible. The stereo microphone (mics/methods) previously mentioned, fall far short of this task and don't look at all like 'us' in capturing ambient acoustic information, but provide plenty of recorded variations that are 'interesting' if not occasionally quite satisfying.

Our hearing reception is neither directional like the Cardioid nor like spaced microphones floating in space. Our baffles (our head) is not a reflective disk (like with the Jecklin Disc) nor made of plastic or foam (like the Aachen and Neumann baffle). In short, all these approaches remain as using methods and materials that have far too little or no 'natural' acoustic perception mechanisms along with a whole lot of 'unnatural mechanisms' by design. How a recording engineer gets the best stereo performance from such mic systems remains the true 'art' that's most discussed. Unless you really need to reject some portion of the ambient (suffering the usual image/sound distortions) considered as interfering 'noise', there is only disadvantage to using direction microphones in any kind of simple or fancy (matriced) array.

The acoustic recording process is only complex because acoustic sound is complex and our recognition of perceived sound is also complex. However, if the (stereo) microphone's processing method is kept eloquent and natural (to us, like HRTF baffled omni pair), then just a simple microphone system is all that's needed to report the complexity of the acoustic ambient to our also complex recognition mechanism via recorded medium.

This is more a situation where the preferred microphone/method does not 'get in the way' of making an accurate (to our perspectives) record of ambient sounds or does not add unneeded confusion with strange ways (to us) of acoustic/electrical signal processing.

Your choice of a suitable omni mic pair to use with HRTF baffle is really up to you as there are plenty of candidates available, some with less than flat response that might be exploited for providing brightness or similar tonal effects. For the most accurate tonality/image, small sized capsules of extended ruler flat hi-frequency bandwidth have a decided advantage. Choices of available HRTF baffles remains much more limited to those shown on my site.

Of equal importance to piano recording is using stereo method recording for other acoustic instruments from flute to drum kit when depth and realism (as you'd hear it) sound is the intention. With this, the ambient you're working in can be a nightmare jungle to a paradise of sound (fill) quality. Using acoustic traps (like ASC's) can help take most of the jungle out of your available working spaces; giving much control of the ambient quality and allowing good results with trying less-than-close mic'd (mono or stereo)
ambient recordings in general.

On the Super 8 equipment, many of the camera repair shops (especially in the larger metro areas) used to keep quite a few of these around for spare parts if for nothing else. Partially reconditioned with some cleaning and new lubrication (very important), these should be available for just slightly more than a song these days.

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Subject: Recording a cappela

From: chris
Date: Sun, 08 Feb 1998 23:02:27 -0600

Hello,
i was just approached to record an a cappela group..but before i
accept i needed to ask a question or two...if i could get some help quick
it would really help me out.
My mic collection consists of these microphones
7 sm-57's
beta 58
1 akg 414
1 rode nt2
2 audiotechnica 4041's
2 beta 52's

with these mics..can i pull off recording 4 singers at once? i mean sure
i can do it..but can i do it well enough ..etc.. and if you think those
mics will work..how do you suggest seting them up?

I ask these questions ..because i am new to recoding a cappela but have
seen the technic of X-Y ing two condensers and having the group stand in
a circle around the mics...i have never really seen a mic slapped on each
person in the studio. what would you ideas be if you were thrown into
this situation???

THANKS
CHRIS

>>
Hello Chris,
You are wise to research this 'most difficult' to record material.

Acappella is a harmonically complex ACOUSTIC sound. The blending of all voices ACOUSTICALLY is what gives this style its unique power to thrill.

Unfortunately, capturing harmony and 'the all-elusive-tag sounds' is extremely difficult with standard mics and methods. Acappella is a unique mix of close direct and a tasteful bit of live 'ambient' that is very fragile to the sound recording process.

The most successful recording method is one that records this sound as we hear it live and not translated by numerous close microphones placement feeding a mixer. In addition, the usual stereo microphones methods & microphones are mostly inadequate to give an acceptable result unless much time is allowed for setup, rehearsal, playback, and adjustment is planned from the start to get even an OK recording.

I know of quite a few that specialize in this style using the patented DSM stereo mic method with easily obtained excellent results.

Visit my web site and go to the Audio Mag. Review section (done by Corey Greenberg last year) to read about how this works. Also, check the multi-track recording article. If this seems of interest, contact me with further questions and good luck with recording acoustic music; there's nothing more satisfying than pure acoustic to successfully record.

Best Regards in Sound & Music Recording,
Leonard Lombardo, Sonic Studios(tm)..."Making Audio History With DSM(tm) Microphones"

<< Subj: Re: b&k

Date: 98-02-10 04:39:04 EST
:
To: GuySonic@aol.com (INTERNET:GuySonic@aol.com)

Leonard,

> I'm not using the old DSM6 mics. which I bought in Dec.1992 at Uncle Stereo's in Manhattan. Generally I tape rock shows though last year I taped a lot of acoustic shows (Springsteen), but normally it's rock shows in clubs or arenas. Only once or twice a year I tape some rock shows at big arenas or the stadium. But I'd say on average I tape 25 clubs, 6 arenas, 2 stadiums. Of course at big arenas or stadiums the bass is much louder. However, that can be reduced when playing the tape back by simply adjusting the "bass" button on the amplifier, but instead I noticed lack of high frequencies with the old DSM6. Normally when I playback my tapes I let the bass button unchanged and instead add more treble.
>
> I'll be glad to get your advice as to what's the best mic. for my needs.
>
> **Thanks**
>
> **Sal**
>
> \>>
>
> **Hello Sal,**
>
> Thank you for advisement on your current requirements. This helps tremendously in determining what DSM mic system should work best. You are using a DAT Deck?.....reminding me of these facts is helpful when memory is not.....
>
> You may have purchased a DSM mic that had a bit too much gain for best loud bass sound recordings. There has always been an education problem with retail sales people. Also, over the years I've learned much more about what workes best for certain requirements.
>
> The DSM-6S/L (Low Gain) is indicated. Using the PA-6LC3 would allow 65 cycle normal Low frequency reduction when venue Bass is excessively loud; use the 100 cycles ONLY for very rare extreme bass heavy sound.
>
> For Medium Bass Rock and PA'd acoustic, the 30 cycle switch allows almost all the bass to be recorded for a full sound.
>
> This should give you all the improvement necessary to allow completely satisfying recording quality for PA'd club & Concert sound venues. This system is still very usable for pure acoustic if close enough and going to higher deck gain settings when necessary to bring

VU peaks to normal.

If you had a MOD-2 mic powering upgraded DAT deck (D100 with MOD-2 is now best available portable), then I'd recommend the PA-6LC $125 (No Switch; 65 cycle fixed reduction) which is used when necessary and not used for other venues that allow the mic to be 100% powered by the deck without the adapter; makes a very simple, extremely compact system with the fullest bass sound possible.

Best Regards in Sound & Music Recording,
Leonard Lombardo, Sonic Studios(tm)..."Making Audio History With DSM(tm) Microphones"
--

<< Subject: Miking Tabla drums?

From: Al .nortel.ca>
Date: Sat, 14 Feb 1998 17:09:36 -0500

Anybody have any suggestions on how to mike Tabla Drums?
How about Marracas (sp?) and other types of unusual percussion
drums. What type of mike and position to use? I tried a SM57
about 6 inches away and got a lot of high end hand noise. I suppose
I should take a more ambient approach then and back the mikes up
a few feet... Is there anybody out there that has actually miked
Drums such as these and gotten a good sound?

--Al
>>
Hello Al,

A medium close mic'd (4-10 foot) ambient stereo will give more satisfying results. Sonic Studios site features the DSM stereo mics and if you could read Corey Greenberg's Audio Magazine review, in the reviews section, you'll get a good description of the techniques to get very natural acoustic instument sound recordings.

I've done moderate close proximity recordings of many types of percussion instruments (single and in large groups) using this technique, but have only at 20 foot distance audience recorded the tabla in the group Oregon; all with very acceptable to excellent results . Exceptional results do depend on a good mix of ambient and direct sound with this technique. The main DSM method advantage is that what you're hearing is going to be what is reproduced from the recording without the usual 'second guessing' prevalent with most other mic techniques.
This saves a lot of necessary rehearsal and occasional real disappointment that often follows from being on the steep part of the learning curve using 'standard' mics and methods in a new recording situation.

Best Regards in Sound & Music Recording,
Leonard Lombardo, Sonic Studios(tm)..."Making Audio History With DSM(tm) Microphones"
======================================

<< Subj: Re: Your Payment Sent .... Cleared Customs today and outfordelivery?

Date: 98-02-19 10:01:56 EST
From: powerpro

To: GuySonic@aol.com

Leonard,

I received the mics today. Thanks for your efforts. Even though they didn't result in me getting them faster, it helped put my mind at ease. I'm very pleased with the mics, and will let you know how the recording turns out tomorrow.

By the way, I plugged them into the mic input on my D100, and played my stereo VERY loud to get an idea of where to start with the level meters. Even when I cranked the levels to 10, the meters didn't go past -12. So, I'd like to ask you for a recommendation on where to start with my levels. I will probably start at around 7, so please let me know what you think. Of course I will use the -20db setting on my D100, and I'm told that this band plays very loud and bassy, so I will set the bass cut at 65, as was recommended by another taper.

Any advice would be greatly appreciated.
Thanks,
Peter
>>

Hello Peter,

Thank you for quickly passing the good news of arrival! Most all home stereos play considerably less loudly than you think because the distortion factor goes up quickly and they just sound loud.

The tact you plan on using the 20 db atten setting and the 65 bass setting seems correct for the 1st time. Your Level knob setting can be all the way full to 10 if needed for desired VU PEAK levels of anywhere from -12 db to 0 db. Best to be a bit conservative (-12 to-6) so that as the band and soundman warm up, you won't be caught short if average SPL levels creep up when not watching.

Thank you for your order and patience. Enjoy!

Leonard
==

Subj: Thank you Will, Received Your order today! Questions/Comments

Date: 02/24/98
To: Bill

Hello Will,

Thank you for placing an order! Your satisfaction is most important and I need to know a little more about your music recording. The DSM-6/M will be a good match for you if you're intending recording traditional to moderately loud (no earplugs required) 'new age' PA'd Jazz. For Rock recording, bands such as Phish, Zero, and what used to be Grateful Dead style will also work well with this particular mic.

However, for the above Rock type bands, bass roll off has never been necessary and using this accessory may reduce the bass too much, specially if not always used in the minimum roll off position of 30 cycles. I realize that some of the larger concert venues of even the soft-spoken Dylan have gotten quite loud lately and bass has also been reported excessive in a few of his venues.... it's hard to tell when bass reduction is going to be needed with some artists varying the presentation with location.

You can give the PA-6LC3 a period of seeing if recordings you are mostly doing are sounding full enough for your tastes.

The Medium sensitivity should allow enough mic output use the (L) setting on the D8 for most everything; this also helps to give a fuller bass sound with Sony Mini-DAT decks.

I don't see anything really wrong with your choice but would prefer for most Jazz and 'moderate' Rock styles to use the D8 with MOD-2 without external powering adapter or bass reduction. More compact equipment and 1 less connection. You can always get the D8 Mic powering upgrade later or may consider going for the quality improved D100 .... only if the D8 has many hours of good use and is getting a bit worn out.

I'll send as ordered to get you started. Let me know if you have any thoughts before I send today or later find you need an adjustment to the original order request of mic an/or powering options.

If being conservative with the D8 VU peaks at about -12 db, your recordings should sound full (adequate bass) and very clean with no bass muddyness (mics not too sensitive).

Best Regards in Sound & Music Recording,
Leonard Lombardo, Sonic Studios(tm)..."Making Audio History With DSM(tm) Microphones"
TEL: 541-459-8839 /\ FAX: 541-459-8842 /\ USA Free: 1-877-347-6642
==

Subject: Video Camera and Mic Question (no DAT content)

Date: Thu, 26 Feb 1998 00:14:14 -0600

What's up fellow dat-heads. I am sorry for the "no dat-content" post but I
figured posting here would get me the best responses on my question.

Anyways, here's the deal...I was planning on taping a local band here in
Austin TX with a TCD-D8 and AT822 mic but I was recently granted permission
by the band to video tape it. I have a Sony HandyCam and I was just
wondering if I could hook the AT822 mic up to the video camera mic-in and
get a good sounding video. Is that all I would have to do or would I need
some other tools?
Please e-mail me back if you can help me out.

Thanks

==
"It's all fun and games till someone loses an eye. Then, its just fun. "
==
>>
Unfortunately, most every camcorder has Audio ALC and NO Manual knob. This is going to make all music recorded with a camcorder compressed and when coupled with a good external microphone, exhibit disturbing 'bass-beat pumping' artifacts.

Use the D8 (using the manual setting!) as the audio recording deck and later sync up the audio with listening to both the camcorder audio and the DAT audio until the 'echo' disappears... (alternately use the pause buttons to get sync) .... both sources will then be in sync .... hit the record button on the VCR master copy deck to record both video AND excellent DAT quality audio **unto a master tape of the performance. Use this copy to do the editing to the final master edit copy; analog video generation quality loss is a factor with this so use the best copy deck available.**

If you're running continuously with both systems when making the recording of the group, then the editing should be less painful. This method also allows the DAT with microphone to be 'in a better place for the audio' and the Video can be then be recorded 'anywhere' where the views are best. This is a tremendous advantage for a quality production of the group.

It's been reported that the sync should stay good for at least 30 minutes once its established. Anyone know about this from experience???

Best Regards in Sound & Music Recording,
Leonard Lombardo, Sonic Studios(tm)..."Making Audio History With DSM(tm) Microphones"
TEL: 541-459-8839 /\ FAX: 541-459-8842 /\ USA Free: 1-877-347-6642

===

<< Subj: Re: : new taper

Date: 98-03-01 21:51:12 EST
From: DMil
To: GuySonic

thank you so much for all the great help.!!!!

can you explain to me WHY I need to use 20db and not 0??? I think if I understand why (layman's terms) I will remember. I taped Steve Earle last night, kept levels between -6 and -12, the accoustics were good, and the show turned out great! I have hope.

diane
 >>
On all Sony decks, the atten switch is actually a mic preamplifier high gain-Low gain switch. Your NORMAL position will be the Low gain or 20 db position which allows 10 times more signal level from the microphone before distortion, much lower high frequency bandwidth distortion, less noise from the mic preamplifierl.

Diane, Do check my web site for these and other tapers tips.

Best Regards in Sound & Music Recording,
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Subject: Binaural Filters?

Reply-To: ian
Date: Sun, 01 Mar 1998 18:36:32 -0700

Hi, I'm new to the list and rather naieve about binaural mics. They all
seem to offer a filter to reduce the extra bass. What is this filter? Is
it just some form of EQ? If it is just an EQ of sorts, could I not load the
recording into my computer, and process it better and precisely to my needs
using a tool like Sound Forge, etc? Or am I misunderstanding what this
filter does? Is it simply a post-process modification or does it somehow
change the way the mics antennuate? If I have the ability to post process
and EQ the recording, would there be any point to getting the filter, other
than convenience?

Thanks, Ian
===
>>
Hello Ian,

Nothing to do with Binaural only the use of mics with almost ruler flat response to 5 cycles!!!!. For grunge shows, bass must be rolled off BEFORE PLAYBACK ON REAL SPEAKERS but can be done anytime during or post.

Real time Bass reduction has the advantage of giving the recording deck more analog circuit headroom that can make a difference if running a bit hot anyway.

My web site has a discussion about using the bass reduction system during a recording and has other taper's tips, a discussion about binaural/DSM differnences, and informative reviews.

Best Regards in Sound & Music Recording,
Leonard Lombardo

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==

<< Subj: bass rolloff ?

Date: 98-03-10 18:21:39 EST
From: Ce
To: GuySonic

i got a question about the bass roll-off.a friend of mine is useing my sonics to record a violin in a church,which setting should he use on the bass roll-off?
any help greatly appreciated
carl >>

Hello Carl,

Normally, no bass reduction would be used.

However, this adapter also powers the mic so he would need to use the minimum setting of 30 cycles to get maximum warmth (although Low frequency content of other instruments would be effected if present), 30 cycles is the best available from this adapter.

If your friend is recording using the Sony SBM-1 processor, then don't use the PA adapter but plug the mic directly in for using the very adequate Plug-in-Power feature on this unit. This would give better warmth to instumental pure acoustic recordings.

If he is using the D7/8, then the powering is much less adequate as the SBM-1 but may be also used directly for good results. The D100 direct powering is totally inadequate without the very necessary MOD-2 upgrade or using the powering adapter.

The bass rolloff benefit (in this situation) would be to reduct trafic motor rumble that is often heard inside city located buildings. Just recording a solo violin would be a situatiion where bass reduction should have very minor (if any) effect on the overal instrumental warmth of the recording. Where motor rumble is very noticable inside, using the 65 cycle position would improve the recording of just violin by reducing interfering noises.

Best Regards in Sound & Music Recording,
Leonard Lombardo

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<< Subj: Re: Stealthing with Neumann KM184?

Date: 98-03-18 19:43:48 EST
From: thomas
To: GuySonic@aol.com (GuySonic)

Len,

As luck would have it, I've been using your DSM6 (or 6S, I'm not sure -- they're on loan from a friend) mics with my D100 (no MOD-2) for the last few months and I absolutely love them when I'm upfront and the crowd is quiet. The problem is I tend to tape at a small bar-style venue (the Station Inn in Nashville) where the crowd can be *very* rowdy and tends to talk throughout the entire show. I've tried taping in various areas in the room, but I've been unable to find a spot where crowd noise isn't a problem (even right on top of the PA).

My only complaint about the DSMs is that I've had to use the 0dB attenuation setting on my D100 because the music is not-very-loud acoustic bluegrass. I know you offer more sensitive mics for those situations, but my primary problem is the excessive crowd noise I get from using omnis in that venue.

So... that's why I'm leaning toward getting some more directional mics.

Tom
>>

Hello Tom,
I'm happy to hear you've got a set to try out, but the powering situation is very poor with the stock D100.

The D100 should be upgraded to provide the optimum power for the DSM as the gain is much less, less clarity, and mic noise is greater with the stock D100 mic power system.

You should actually be using the 20 db setting if at all possible (even if the level knob needs to go to full up) as this is technically a better preamplifier front end with adequate headroom. All PA'd sound recording should be done in this setting for audibly better results if you can manage it.

You would get much more enjoyable tapes if using a non-directional mic (like the DSM headworn in some fashion as suggested) for shows where the audience and performance are more conducive.
Getting a directional pair may help in this one place and for the reason you described but will give much lower quality recordings in general specially for better venue situations. I have found that bar crowds are using live music more as wallpaper than as a main interest; this seems to be also happening more at concerts I've lately attended. Headwearing the DSM mics helps give a directional aspect to this interference that is more easily ignored like at the live venue, but trying to remove it with directional mics will also take a major portion of the ambient stereo sound quality away, increases distortion, and loss of that nice bass warmth full sound typical of the DSM.

Perhaps recordings in this one place is not a good reason to get poorer results in general as is expected with directional mics of modest cost used for stealth setups.

You must decide what's most important and most vendors will give you 30 days to get a refund in order to try out a set completely; ours is 60 days.

Please visit my web site (listed below) for a lot of info on mics (with mag reviews), taping, and tips.

Best Regards in Sound & Music Recording,
Leonard Lombardo

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==

<< Subj: Re: mics

Date: 98-03-20 04:54:43 EST
From: TeRRy
To: GuySonic@aol.com (GuySonic)

> I have just gotten a Sony TCD D100
> I am interested in recording some hard rock concerts.
> Could you tell me what you would recommend for this?
> I plan to record both inside & outdoors.
> Also, could you give me a cost on those items?
> Thanx
> TeRRy

> >>
>Hello TeRRy,
>If you could check my web site first to get a feel from the info and reviews,
>this would help in answering the other questions that are important to you.
>
>The DSM-6S/L or EL (lower gain models) with or without the PA-6LC or PA-6LC3
>powering adapter for powering and excessive bass control may be the best
>approach to hard loud bassy type music recording. Otherwise, the D100 can be
>upgraded to power the mics fully if bass reduction is not necessary and would
>make for a simpler system.
>
>Best Regards in Sound & Music Recording,
>Leonard Lombardo

I did go to your web site already and spent about 3 hours checking it and the links out. Although it is very informative, I am afraid that I just do not have the technical knowledge to know enough to make an informed choice. I was hoping that you might be able to figure out a system for me to use with the TCD D 100 that would be the best to capture rock music performances. I do not mind spending a little extra to get a better sound, up from the basics. I did not get the super bit map adaptor from Sony, as I have no idea if it would be worth it. What do you think would be the best way for me to go? If it would help, I could give you a call to talk it over. I want to get a good system in time for the summer shows.

COMMENT: The recording performance of the new D100 DAT is almost identical to any improvement gained with using the SBM-1 with the older D7/8 Sony decks. You've already got an optimal system with just the D100!
Rock music is pretty varied in loudness and sound quality depending on what groups are your interest and location of the venue. Groups like Phish, Zero, and most of the Grateful Dead concerts I've attended seem to be of moderate loud volume but also have high quality sound balance (not excessive in the bass). Other music groups of Blues/Rock/Raggae style (specially in small clubs) can get very loud where earplugs often are desired and bass is purposely excessive and/or the hall is boomy to produce major bass loudness peaks; this is where the bass reducing PA-6LC(3) adapters are an advantage to get an even sounding recording that's playable on normal stereo equipment at more reasonable loudness.

Also - in the last few shows I have tried to record with my ECM-959A mic, I get about 20 minutes of the show, and then it just stops and becomes a low volume clicking with NO music at all. I was not using the 20db setting and was on #6 usually around front row center. Any idea what is happening here?

Talk to you later
TeRRy

+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+
The lover of life's not a sinner
The ending is just a beginning
The closer you get to the meaning
The sooner you know that you're dreaming

*** Ronnie James Dio ***
+=+
>>

Hello TeRRy,
The problem with getting no sound seems either like a poor connection (pehaps the mic plug slipped out a bit) or the battery in the mic went down? DAT decks are usually not intermittant and either work or don't work. The mics and connectors are another story.

While your Level Knob Setting of #6 indicates a safe input signal (atten switch) condition, but being also at 0 db when using the 959A seems to indicate either the mic's output is abnormally low (very weak battery) and/or you also had the deck's ALC setting switched on (which defeats the manual Level Knob control entirely).
You should use only the manual (switch) setting and mostly always be in the 20 db position for moderate to loud source recording with most small & battery operated microphones. Check to see where you have these switch settings on the deck's back.

For a DSM microphone system, it seems that the DSM-6S/L (low gain) set $(450) should be a good fit. Having the D100 mic powering upgrade ($85) would allow a simple, reliable system, and full bass frequency performance for doing most venues with excellent to slightly below average quality sound.

If (with the above system) you're finding that some or a majority of your recordings are too bass heavy (this depends on the group's music style and venue locations you frequent), then the PA-6LC ($125) mic powering and bass reducing adapter can be plugged in-series with the mic and used only for those occasions where bass is excessive and/or the room is boomy.

If you already know that very loud bassy/boomy sources are where you're mostly going to find yourself recording, then forget the internal mic upgrade on the deck and just go for either the PA-6LC (fixed 65 cycle setting) or the more versatile PA-6LC3 ($200) with its 3 switched 30/65/100 cycle bass control settings. The PA-6LC3's 30 cycle position allows fuller bass recording (although it's still being reduced) for when venue sound quality is better.

Perhaps a call is warranted if more issues need to be discussed?

Best Regards in Sound & Music Recording,
Leonard Lombardo
==

<< Subj: quick question...

Date: 98-03-21 14:23:55 EST
From: wmaangel   net
To: GuySonic@aol.com

Hi, I am considering buying a pair of your DSM6 mics, but wanted to ask you a quick question, because I may not be understanding something...
I am thinking about getting a D100...OK, if I get the PA6 to power the mics, can I then go "line in" on the deck?
Your answer is appreciated, I have figured out that if you get the "mod2", then you have to go mic in for "plug in power" but I just wanted to be sure about the line in ability with the battery box..

Thanks!

Matt C.
>>
Hello Matt,

DSM microphones have much lower output by desugn than most the tiny other mics and will only work with mic inputs; makes it easier to know where to plug in and setup over other mics that use both line & mic inputs.

You will not be overloading your deck's preamplifier (with the DSM) and will be using the D100 in the 20 db attenuated positon for most everything with excellent results.

Having the internal mic powering upgrade or using the external power system is a decision that's more easily made knowing what styles of music are your main interest. Bass controlling powering adapters (PA-6LC & PA-6LC3) are necessary for best results for recording very loud, excessively bassy and/or very boomy bass. If the majority of recordings are of this type venue, then the external adapter with bass control is better for your requirements. Otherwise, the internal MOD-2 Deck upgrade for Phish, Zero, ABB, and Grateful Dead type loudness levels and sound quality is a better choice for most tapers.

It seems that you've already seen the Web Site listed below for detailed info?

Best Regards in Sound & Music Recording,
Leonard Lombardo

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===================================

Subj: Re: DSM-6/EL mics

In a message dated 98-04-16 16:49:17 EDT, you write:

<< Subj: RE: DSM-6/EL mics
Date: 98-04-16 16:49:17 EDT
From: larz
To: GuySonic@aol.com ('GuySonic')

Another thing I forgot to ask you was regarding the audience noise etc,
obviously in live rock concerts, the crowds around you can get quite noisy,
and as such, cardioids could be better at eliminating all that unwanted
noise. Do you have anything in the way of cardioids or do you feel that your
binaural mics do a pretty good job of that themselves?

Adios,
LarZ
>>

Hello LarZ,

Yes it would be nice to mute out that motor mouth or Yahooo near you when it happens. However, the mics that suppress side sounds also suppress a lot of what makes for really good full sound steeo recording. Your best tact is to get some distance from some people or get closer where the sound is greater than the closest 'party' noises.

Cardioid mics do suppress some of this but you'll be much less happy with most all your recordings using this type of mic exclusively ..... its mostly a special case microphone. However, if your music tastes ALWAYS places you within a riot/party zone of which you cannot tolerate as part of the natural ambient 'scene', then maybe the lower quality recording with side cancellation is a better choice for you at this time. When recording other venues where the crowd is really into the music more that the party, then the DSM will be much more versatile and most satisfying over any Cardioid set. There are some rather costly ($1200 a set and up.... and larger size) cardioids that actually sound very good for being directional. I hope you don't really need directional mics......... you'd be a lot happier with this hobby for getting really exciting results with using omni headworn mics.

Best Regards in Sound & Music Recording,
Leonard Lombardo

<< Subj: Love for music!

Date: 98-05-03 10:25:48 EDT
From: Anders J
To: GuySonic@aol.com (Leonard Lombardo)

Hi Leonard!
As you might remember I bought a pair of DSM-6S from you half a year ago. Just to make a long and very nice story short: Music is very important for me and these mics with the D100 have brought me closer to music than any other HiFi gear have done for me!!!!! I am right now listening to a solo piano concert i taped in a church here in Stockholm this weekend - fantastic - the dynamic of the grand piano and the sound of the church room is all there!!
Well I could write much more to the praise of your products but let's go to my questions:
- When I record acoustic music (like this piano concert) I set the level at 10 on the D100 (pad -20dB) but still don't get more than -20dB level on average. I don't want to take the mic pad away - this only adds noise and I am afraid of mic preamp distortion (right??). Would it be a good idea to get a new pair of mics with higher sensitivity? Though I have been taping very loud music, I have never used the line in - the mic input on the D100 has been insensitive enough.
The sound quality of these low levels recordings are very good and the noise is no problem - but still a somewhat higher level shouldn't hurt - what do you think????

- Quite often I don't find the "mics on the glasses" to be the best way to wear the mics. Instead I have cut small holes on the sides of a cap, just in front of my ears. The cap is black and from some distance you don't see the mics. Next option is to take off the cap and place the cap on my knee. I sit on a chair and cross my legs and place the cap on the knee pointing towards the sound. This works very well for me. But - do you have any other ideas regarding mic placement??

BTW - thanks for you very nice web site!!!

hope you have some time to answer my questions!

thanks again!!
Anders
--
It's the latest gigg I ever played in my whole life
- but what the heck - here we go! 1994 M.S.
>>
Hello Anders,

Thank you for sharing your appreciation of the DSM recording system.

You are safe to use the 0 db position.......... your level knob will probably set around 5 and if at #4 or above indicates a safe input condition against overload.

Your cap on the knee seems a workable situation........ stereo image is less ideal though.... cap placed on the head is more ideal.

A DSM-6x/H or EH would be better suited for pure acoustic......... DSM-1/M is ideal for just acoustic/nature where SPLs are less than 110 db spl......... no heard mic or deck noise but quite limited to doing just this type of recording.

Best Regards in Sound & Music Recording,
Leonard Lombardo

Subject: M1 question--bass distortion

Reply-To: h.net
Date: Sun, 03 May 1998 13:52:23 -0500

I used my new M1 in a stealth situation for the 1st time.

I recorded Ziggy Marley with Sonic Studios and I've got distortion on the low end. The levels peak around 3db, the highs and upper mids sound great but the bass is way overloaded.

Is there anyway to repair this?

The mic attenuater was set on 0db, I didn't think the music was that loud.

Anybody who can help and wants a copy of teh tape will get it free of charge! It was the best ZMMM show I've ever experienced.

Jai

  >>
This is an unfortunate and non-recoverable error (distorted content can be reduced with bass rolloff but not eliminated) in deck setting.......... using the deck in the -20 db attenuation setting avoids this type of mic preamplifier overload .... the mic is not overloading, ... it's the deck's mic preamplifer causing the recorded distortion.   VU levels will not show this condition............ the LEVEL KNOB does give an indication of likely overload when positioned below the #4 mark....... #4 to full up #10 (for desired VU indication of about -6 db VU) indicates that the deck's mic preamplifier is being operated in a safe-from-overloading setting.

This and more tapers tips are located on my web site listed below.

Best Regards in Sound & Music Recording,
Leonard Lombardo

<< Subj: Mics: good news and bad news

Date: 98-05-05 18:21:32 EDT
From: Gary
To: GuySonic@aol.com (GuySonic)

I did the DV video shoot with the DSM mics and the attenuator you suggested, and it came out well. The attenuator definately did the trick.

I did have a little trouble with the 1/8" mini-plug I used:
when you wiggle it, it makes noise. I noticed that the plug you use on the mics is noise free. My plug was essentially designed for headphones, do you know if there's a slight difference in the spec, or a difference in the quality, or are some brands of miniplugs just better than others for low-voltage signals?

I do have one problem I hope you can help me with. A crowd surfer kicked me during the show and now one of the mics is making a slight high-frequency buzzing noise. Wiggling the cable at the mic end can make the buzz softer. Have you ever heard of a problem like this? Do you think you can repair it?

I'm sure you usually charge for this sort of work, but I am basically unemployed and a bit destitute. I would, however, like to write up a report of how I used DSM mics to solve the problem of distortion with Sony DV cameras. If I post this report to at least two mailing lists (Dat-Heads and Digital Video List), with a published article possibly in the future, could you look at the mics for me no-charge?

Also, could you please tell me the model number for DSM mics with an
inline power supply and low output, and also, do you sell an
attenuator so people can purchase an "off the shelf" solution?

thank you!
--Gary
>>

Hello Gary,

Follow the maintenance tips on my web site ..... "mini-connectors ins' and outs" to clean the plug and jack to eliminate noises from dirty or corroded metal mic connectors. Hum can be from a poor connector grounding connection or a soft failure of one of the capule's grounding connections......... locate the pickup channel with the hum and give the capsule a mild squeeze with thumb and forefinger (listen on headphones) until the hum disappears with the 'resetting' of the capsule grounding connection. This fix will work for only a short time (day, week, months) but will reappear again .... fix with a slight squeeze before using. There is no permanent repair for this other than replacing that mic channel completely with another microphone assembly...........cost is $150 and is possible only when an accurate match is found for the good remaining channel pickup. Mic can be traded up for a newer unit at 30% off new mic price.

In-Line -15 db attenuators are available for $60 and come with a choice of mic connectors (mini, 1/4", and XLR).......... usable only after the powering scheme of the mic used........ not between the mic and the mic power.

Best Regards in Sound & Music Recording,
Leonard Lombardo

Subj: Re: battery drain

Date: 05/06/98
To: .com

Hello Tim,

You've a few questions posed so I'll reply within your sent text.

Best Regards in Sound & Music Recording,
Leonard Lombardo

TEL: 541-459-8839 /\ FAX: 541-459-8842 /\ USA Free: 1-877-347-6642
<<<<<<<<<
In a message dated 98-05-06 13:47:14 EDT, you write:

<< Subj: battery drain
Date: 98-05-06 13:47:14 EDT
From: tjp@.com (Tim P)
To: GuySonic@aol.com

leonard,

i borrowed a set of dsm-6 mics over the w/e to record harry connick jr. my friend sent them w/ a battery in the pack. i assumed they would not be draining juice of the battery since they were not plugged into a recorder.

i did a short sound test w/ the supplied battery on music from my computer speakers. the levels wer were low, so i assumed it was due to the low volume. i get to the show i get the tape rolling and still low levels (i am almost puking). i write the incident off and hope to do better the next night.

the next day i put a new battery in and test the same as i did before. WHAM. great levels! i almost hurl, since it was the battery that was almost dead and i got no sound :(

did the mic wear down the battery during shipping and waiting several days for the show? or is it more likely that the battery was already dead upon arrival?

REPLY: The battery will last a continuous 2 months if left in the holder or with the mic pluged in to the Powering adapter........... most just unplug the DSM-6 microphone from the PA-6 powering unit.......... your friend sent you a very used battery or one that was left connected for over 2 months. The much older DSM-6P sets had the battery powering unit hardwired to the microphone.... a one piece system .... removal of the battery was the only way to stop power consumption.

btw the recording w/ the fresh battery was great. just want to kick myself in the ass for not checking another battery.

questions about the purchase of a set of your mics. 1) for the dsm-6 you use the panasonic capusal, right? why is the dsm-9 $1800? it uses a b & k, right? can the b & k capusal be that much better? i ask this because, from your webpage, it shows that the dsm-6 has a slightly greater range than the dsm-9. 2) what about the 'signature' series? what does that add besides $100?

REPLY: The DSM-9 capsules are very costly to purchase and much more difficult to get a decent match....... they also require extensive mechanical/acoustic modification to remove the coloration prevalent in the stock microphone.......... they are worth the money but, not really that much better than a DSM-6x/x model fitted to your sound/music recording requirements. Actually, the DSM-9M has more ability to handle medium to extremely loud sound recording......... the DSM-6x/-has 5 ranges of sensitivity sub models to handle full recording dynamic spectrum where the DSM-9 can cover the necessary low noise operation AND SPL loudness max. range of 3 models (M, L. EL) with quality performance.

i had graet success w/ the dsm-6, but am nervous when wearing the larger mic compared to the dsm-9 or as the core sounds appear smaller w/ the pix on their web page w/ the mic compared to a dime. i mention this, because i am in a wheelchair and ushers are always hanging around where i sit or they escort me to my seat at events or asking if i need any assistance.

my uncle came up to me at the show and and in 3 seconds noticed i had the mics on (he had no idea i was going to be there, let alone taping). this w/ lights on and a few minutes before the show starts. i normally wear glasses, so i already had the mics on and ready to go.

honestly, it would take me a long time to save for the dsm-9 mics. so i am not sure how much of a potential cutsomer i am for the dsm-9 mics.
sincerely,
tim parrish
-=-
Tim P/

"When I make a promise, you can bet that it's true
So put your chips down, baby and empty your pockets, too."
-- Don't take your love away
-- Neil Young

>>
Hello again Tim,

Moderate length hair (just covers the ears and sideburns will hide the microphones on eyeglasses..... pickups secured (with the metal slider) at about the sideburn area. Other placements are within the sweat band section of a backwards worn black or very dark colored baseball cap (hides the cord exiting down the back) through a few specially made 'button holes' large enough to let the pickups protrude without mechanical interference or rubbing........... the loops and cord can be duct taped or sewn secured. Headwearing the mics does work the best for stereo sound quality but some users have place button holes within the colar of a shirt or vest position the mics close as possible to the neck......... this is a difficult area to keep mics from getting rubbed or bumped......... much greater care must be taken to avoid this when mounting mic within clothing.

The DSM-9M mics are not $1000 better .... don't feel shorted by not affording them......... much satisfaction with a DSM-6x/x is most certain.

The S designation is a best channel matched set and can be audibly better depending on what you've learned to listen to and the quality of the source.

Cleaner bass and more accurate ambient imaging are some of the advantages of going the $100 more for this premium matched grade.

Best Regards in Sound & Music Recording,
Leonard Lombardo

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<< Subj: A question?

Date: 98-05-07 23:56:22 EDT
From:
To: GuySonic

Hi,

Tried out the new mic's and am very happy with the sound.

One question however: Would you recomend any alternative ways or places to mount the mic's(other than glasses), with the purpose of being more secretive(concealing them more). Such as on a hat, etc.
And how would it affect the sound.

Thanks for your time,
Joe
>>

Hello Joe,

Here is a reply to a similar question:

REPLY: While the DSM mics are a bit larger (due to an advanced acoustic/mechanical system found in no other mic system), many users more concerned with visibility than stereo image quality have opt'd for other lower profile mounting methods.
Reversed baseball caps are popular as the 'bill' hids the backside cord exit and the cap is useful to many of at least my age group with a lot less hair or with balding spots. A specially made 'oversized button hole' place in the 'sweat band portion' of the cap .... somewhere forward of the ears & allowing just the mic portion to protrude out and the mounting loop/cord secured with duct tape or sewn to run inside the sweat band to the cap rear (or the bill area if reversed).
Another method is to use the button hole method on a selected shirt or 'recording vest' that completely hides the wire with another access single large 'button hole' (large enough to thread both the mics/cords through) under the rear section of the collar so the cords are invisible as they run down the inside backside of the garment to exit at the trousers belt area for deck input access.
When pickups are either hat or collar mounted near or to still have some major body mass (head or neck) is directly adjacent and between the mics, the stereo image is still quite good.

Best Regards in Sound & Music Recording,
Leonard Lombardo

<< Subj: Sensitivity vs. noise floor

Date: 98-05-08 10:31:43 EDT
From:)
To: GuySonic@aol.com (GuySonic)

Does the noise floor vary with the raising of the sensitivity of the DSM
mics?
If so, how much between the DSM6s/m & the DSM6s/h?
Thanks!!!

Sorry for not asking all my questions at the same time. I'm still very new in this outdoors recording field...

God Bless!!!
David
>>

Hello David,

There are two mechanisms that are at work with going to a more sensitive mic........ 1st, the higher output of the more sensitive rated mic does help in better S/N performance of the mic itself........ by 1-2 db (DSM-6x/M-H), ... 2'd, higher mic output also gives more S/N performance from the deck's mic preamplifier by supplying more signal in relationship to the deck's mic preamp stage self-noise. The quietest DSM set is the DSM-1x/H,M,L models that give 5-10 db quieter performance but have limited loudness handling............ not really suitable for amplified or moderately loud sounds.

Best Regards in Sound & Music Recording,
Leonard Lombardo

Subj: Re: order info.......why DSM-6S/M may be a better choice

Date: 05/12/98
To: endo
In a message dated 98-05-11 03:55:18 EDT, you write:

>What kinds of non amplified sounds would you be recording....... piano, Jazz
>music, Classical, Vocals,....... ?? Would the Rock music sounds be not so
>loud as to make the ears ring very much afterward? I try to suit the mic
>closely to your tastes so that the very best results are easier for you.

Thanks of your kindly advice.

My target recording sorce are....

a) Rock music without "ears ring" (Not so laud).

b) Non amplified sounds.. accodion, guitar, vocal ensamble,
in a very small room (usually until 40 peaple).

I want use AGC in D-100, My frends DSM6s/L is good without mic. att.
if the att. set -20db pritty low revel.

Then if I select DSM6s/M and without mic att. in D-100,
analog circit clipping probrem appear or not.....
I afreid this probrem.

Do you think ?

Yasuhiro

, university of Tokyo
>>

Thank you for relating more details on your recording interests.

I'm thinking that if you're main interest was just the acoustic music
, then......... the DSM-6S/H or EH (high sensitivity or even Extra High) would be much better suited for just this.......... it would allow you a chance of using the D100 with the -20 db attenuator ...... better performance and overload protection is found in this position if enough signal comes from the microphones....... the sensitivity of H and EH is 5db to 7 db more than the model you borrowed from your friend who records just Rock Music. However, the need to record amplified music and louder Rock must temper this suggestion to also include this ability .... this is why the DSM-6S/M with 3-4 db more sensitivity/output signal may work best for both.

The 3-4 db higher output of the Medium gain microphone could allow you to sometime use the -20 db attenuator with acoustical but not as often as with the High or Extra High sensitivity model. Overload of the mic preamplifier is also more likely with the Medium Sensitivity mic selection (as you've anticipated) when the deck is set to 0 db...... as you are now doing with the borrowed set........ when the acoustic music is loud or when positioned very close.

A way to avoid overloading the deck with signal from any microphone is to notice where the Level Knob is set for getting -12 db VU to -6 db VU........ if the Level Knob is at #4 or higher....... to even #10 (full up level), this indicates there is little chance of overloading the mic input.... when the knob is positioned below #4 for the same VU readings, then then indicates there is a chance of overload and using the -20 db attenuation and turning the Level Knob towards full up is suggested.

Let me know how this appears to you.

Best Regards in Sound & Music Recording,
Leonard Lombardo

<< Subj: First use of DSM-6s

Date: 98-07-06 18:49:46 EDT
From: Aaron
J Reply-to: Aaron J
To: guysonic@aol.com (guysonic@aol.com)

Leonard,

I received my TCD-D100 and the DSMs plenty of time before the Joe Satriani concert July 4th (10-11:30pm). Thanks again. I learned a lot recording that outdoor concert, like show up hours before the start and include a Mini-Mag flashlight with your kit.

Unfortunately, I arrived about 50 minutes before the start and was only able to position myself about 10 rows back directly facing the right side PA stack (and only after talking some people into letting me stand next to them). It was completely packed.

For the most part, the recording is truly spectacular. There are points where you would swear I was patched into the board. We were blessed with an excellent FOH mix except in some instances where sequenced material was played back. One of my friends says my recording of Summer Song is better than the one on Disc2 (live) of the Satriani's Time Machine 2CD set.

I do have a couple of questions based upon this experience, though. First, I'm wondering how the mic limiter (not ACG, just limiter+manual) responds with the DSMs. I noticed (after listening to recording) that I fooled with the level too often because I didn't want to use the limiter. It wasn't entirely my fault because some of those sequenced tracks really cut through too loudly. From what I heard playing the tape back through my Carver AL-IIIs, the ideal overall level seems to reside at -4dB with peaks consistently going to 0 to get that polished CD sound. Will the mic limiter allow me to safely raise the level average from -12 (recommended in Sony manual) to closer to -4?

REPLY: The limiter seems to go into action (just like an automatic level knob) at about -3 db to limit levels from going much over -2 db......... this action can be a bit too audible (in the form of dynamic compression or 'down pumping') especially if you've manually advanced your level knob for over -6 db average levels with the limiter acting on the peaks frequently or even constantly.
Because of the great variations during most live performances, it's best not to try to push average levels much greater than -12 db VU............ allowing the peaks to approach about -6 db VU will give you more warning when the program starts exceeding previous averages............ going for that 'polished' CD level is too difficult to achieve unless having much prior knowledge of a certain venue's range of loudness........... best to be conservative and get a much smoother sounding tape without overloads or frequent need for manual level adjustments.

Secondly, in certain places in the recording, you hear these funky phasing anomalies where the image bounces from left to right and vice versa. The phenomenon kinda sounds cool through headphones, but it's annoying when you hear it through loudspeakers. (I positioned the mics very close to the hinges of my glasses, not back near ears.) Was this caused from phase cancellations because I was standing in front of the right side PA instead of being centered between the two stacks? Maybe the wind, or me moving my head? Interestingly, I don't recall hearing them when I was taping.

REPLY: While wind currents can shift the balance (a little bit) and especially the loudness when at a moderate to far distance from the stacks, most shifts in balance are from left/right head movements............. consider you've also attached a video camera to your head........ with your head direction controlling the pointing of the camera view........... the stereo is similar in effect with turning your head.......... knowing this will help you make unconscious hard left/right head turns. Looking up and down will not effect the stereo .... but, can cause the high frequency sounds to vary if sounds are being blocked by those directly in front of your view of the stacks.

Lastly, I had terrible luck with battery life. Thankfully, I have 4 pairs of NH-D100s, but I didn't even get 1.5 hours from the first pair and they were fully charged! I turned the backlight on and off a number of times to check levels, but that shouldn't have cut my bat life in half. Any ideas? (Did not monitor with headphones - too loud anyway.)

REPLY: I've heard similar reports of short battery life with those NiMHbatteries................ the light is not a problem with recording time....... the NiMH battieries can be quickly damaged if inserted too soon for charging again (before being mostly run down) or for a second time on the charger by accident. I've only used one set of these........... when new and before charging, I placed in the deck and ran down........ took just 20 minutes........... then placed on the charger for 5 hours........ about 2-3 hours longer than when the green light showed (this will not hurt the battery ..... as long as the red light doesn't show a second time).
It doesn't matter if these batteries are stored charged or discharged.......... no memory effect........ just sensitive to overcharging....... and they run down a bit faster when not being used than the NiCad types.
This original battery gave me more than 3.5 hours recording time........ it shows it's possible .... but not as reliable as using an external BC-1 battery pack or the L91 Eveready lithium cells available from Radio Shack for two @ $6.99........... these cannot be recharged but will last for at least 5 hours ..... then quit without much warning.

Any suggestions to further improve my techniques is greatly appreciated.

Thanks again,
ajc

Aaron J.
>>

Subj: Re: First use of DSM-6s
Date: 07/07/98
To: ajc@execpc.com

Hello Aaron,

It sounds like you did better than just OK with your very 1st time using the new system. I'll attempt to answer your questions within the copy of your message.

Thank you again for deciding on using DSM mics............ you should be a very satisfied recordist with just a little more experience.

Best Regards in Sound & Music Recording,
Leonard Lombardo

Sonic Studios(tm)..."Making Audio History With DSM(tm) Microphones"
TEL: 541-459-8839 /\ FAX: 541-459-8842 /\ USA Free: 1-877-347-6642
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<< Subj: Re: question

Date: 98-07-06 17:32:56 EDT

From: TheKind
To: GuySonic

Thanx for replying. Right now I have a Sony MZR30 Portable minidisc recorder. But I plan on getting a Sony D8 by the end of the summer. Most shows that I tape are heavy metal type bands that play in smaller clubs and theaters and some arena shows, but mostly smaller clubs and theaters. I've noticed a lot of people talking about Mod1 and Mod2 for Sony DAT recorders. What exactly is this? Would i need this if i got your mics. Plus do your mics come with a bass reduction filter, or is it not necessary with your mics. Right now i am taping with Sound Professionals Binaural Mics witha bass reduction filter.

let me know
brad
>>
Hello Brad,

You can plug the DSM-6S/L or DSM-6S/EL (Pro Low Gain or Extra Low Gain) mics directly into your MD and power it with the mic powering feature standard on your deck........... the MOD-2 is an increase in the power available stock..... gives best performance from the DSM mics ...... but, is not absolutely necessary with the MZ-R30 which has fairly good power available........ However, the need for bass reduction control for good listenability of Heavy Metal recorded live Rock is often desirable.

For optimum powering the mics AND bass control, there are several choices of PA-x adapters that work in-series with the mic. These are shown (only until I get the PA page completed) listed on the web site home page. The PA-6LC2 is a versatile model that allows 65 cycle bass reduction (the most usable frequency of control) and can be turned completely off for full bass recording when conditions allow.

Best Regards in Sound & Music Recording,
Leonard Lombardo

<< Subj: Your Mics.....!!!

Date: 98-07-12 08:20:09 EDT
From: TeRRy
To: GuySonic@aol.com

>Yes it did come and the mics, adapter, and cleaning kit are in the mail
>Saturday to you........ let me know how it's working after a few experiences.
>

>Sure will..........Thanx a LOT!!!
>TeRRy

Just thought that I would drop you a line to tell you how HaPPy I am with the Mics that I got from you. I was a bit concerned about spending so much on a set of mics, but I am EXTREMELY HaPPy with the results that I have been getting with them. Basically, I can just say that when I play back a tape, it is like I am right there at the show again.

Also your sheet of recording tips has helped me tremendously.

There is one problem that I have run into......

At a recent Metallica show, I was about 30 rows back and although there was decent sound, the recording had a kinna echo reverb sound to it. This was at an outdoor venue. Also, I had the misfortune of being near some little bitch that saw me start the recording, and from then on, she started to scream at the top of her lungs, almost every 30 seconds. Her damned screams come thru better than the show did. I was recording at #6 with the -20db on. I could have turned it up a bit, but noticed too late to change the settings. Usually I am MUCH closer and have EXCELLENT results. But for those times that I am unable to get those closer seats, is there anything I can do to get a better recording with way less audience in it???

Aside from that, show, I could not have gotten a better recording unless I was plugged into the sound board. Those mics are GREAT. Thanx a LOT!!!

TeRRy
>>

Hello Terry,

Thank you for the update on how it's going with the new system. Your response to the enjoyment of recording a good show and later listening to your experience is one I wish to be more common............... until then...... please spread the good word about your new hobby to appreciative friends.

As far as the audience noise, it's always best to try to not let those around you know you're recording.............. I've had similar problems with some of those close by getting even more obnoxious than normal when they know they're going to either be on a recording....... or......... feel ambivalent about your personal taping of a show that they feel personally not to have guts to take the risk to attempt.......... mostly for fear of getting caught doing a 'no-no' if anything .... or .. it's their reaction to seeing you doing a 'no-no' in their opinion..... supported by the venue policy no less.

There's no technical cure to solve this kind of interference............. only a personal decision to try to deal with the person and the over reaction to your taping......... this is a personal area............ you can either move to another location, tolerate the situation,......... or......... try to deal with whomever .... on a personal level that gets the desired result of them toning down the noise to tolerable levels.

If they don't know you're taping........... you can appeal to them as not being able to enjoy this, 'your very favorite group', very well with them carrying on in such a manner............ could they be kind and thoughtful enough to try being more thoughtful or whatever seems appropriate........ don't try the angry approach if at all possible...... but, sometimes you must be persistant enough to make it happen.

If they know your taping the show, try the same tact, but tell them you'd be so grateful as to send them a copy........ which you'd never do normally to anyone....etc........ if they would please try to make it a tape of mostly the music without all that constant screaming ....or.......... whatever seems appropriate to the situation.

Be as humanly kind, but also as compelling as possible............ push aside most of your annoyance, calm any anger .... what

those around you are doing that risk you tape is not personal......... your in a situation that is part of learning experience in general .... it's hard to tell the motivation of that obnoxious acting individual next to you............... I've had this experience numerous times......... and it's hard to tell the devices that some people use to get 'your' attention and from those of seemingly hostile or anti-social behaviour.

Good taping and hope we all may improve those 'personal/social relationships' and as the need arises.

Best Regards in Sound & Music Recording,
Leonard Lombardo

Sonic Studios(tm)..."Making Audio History With DSM(tm) Microphones"
TEL: 541-459-8839 /\ FAX: 541-459-8842 /\ USA Free: 1-877-347-6642
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<< Subj: No Subject

Date: 98-07-26 13:51:38 EDT
From: Kevin
To: GuySonic

Hello -
I am very happy to have found your page through a friend who is a very satisfied customer, Paul Hostetter of Bonny Doon, California. I am writing to ask some advice. The projects that I need equipment for are live CD recordings of acoustic duets - instruments include fiddle,piano, guitar, flute, clarinet and irish bagpipes. I will also do some vocal recording - storytelling. What rig would you recommend? Looking forward to your reply
thanks,
Kevin Carr
>>

Hello Kevin,

I appreciate hearing that Paul is doing well and happy with the results of his recording .... I haven't heard from a friend of Paul's for quite some time ...... Welles B. Goodrich ..... Welles is THE A cappella expert ..... maybe ask Paul about him and let him know of my web site listed below.

The web site has listed many models of DSM mics, but your kind descriptions of your recording intentions helps me greatly to make a first suggestion
to appropriate models.

If you are needing a complete Recorder & mic sytem, the following should be about perfect for you:

Sony PCM-M1 or TCD-D100 DAT deck ..... upgraded to correctly power any DSM-1 or 6 model mic $850.-
(Models are virtually identical.....but M1 is black cased and the pro version, without SCMS copy system and without remote controller-display/earbud phones included)

DSM-6S/EH microphones..... very precision matched ..... Extra High Sensitivity .... excellent for a wide range of acoustic instrument and voice recording ...... Special Priced right now at $450.- (reg. $550.-)

BC-1 .... external D100 or M1 battey system .... allows over 25 hours operation on 4 alkaline C flashlight cells .... makes the deck much more practical to power reliably .... with belt/shoulder strap case for also carrying the deck $80.-

If anticipating recording outdoors .... where winds are likely .... the DSM-WHB windscreen headband ($125.-) might also be considered a necessay accesory for this type of recording .... Using a generous sized open umbrella......(the highly curved & deep 'dish' type) .... to block 'predictable' wind directions can also work with your head well placed inside the wind quieted umbrella interior.

Ordering details are (on the Ordering page) on my site listed below .... as well as most details on the products discussed ...... (still need to get the DAT deck details shown).

More questions welcome anytime......... stock is available for immediate delivery on the products discussed.

Best Regards in Sound & Music Recording,
Leonard Lombardo
========================

<< Subject: Piano and Voice, Disaster.

From: "Alisdair " <.com>
Date: 31 Jul 1998 07:07:12 GMT

I have had to record a clasical piano and voice on only two tracks. The Piano in mono, well seperated from the voice on the other track. I was desperate to gt the grand piano in stereo, but my computer threw a tantrum, and I had no option but run with it. So any ideas what to do with this piano?

In sound forge I have created a stereo file, and from there I can make a pseudo stereo track. It gives the piano width so I can put the voice in the middle. But is there a better solution. I am prepared to hire, buy, whatever it takes.

All ideas gratefully received.

Alisdair ,
England
--
>>
Mic the piano and the vocal with a two matched omni mics.......... the DSM mics are matched but are best when used within the scope of the patented method of recording stereo. With this method, the vocal is centered and in front of the piano .... everything is acoustically mixed and will sound natural and very wonderful....... mixing sounds as complex as piano and voice in the manner you've attempted will give you a headache and with sound that can cause headaches for others who try to enjoy such.

See my web site and reviews there (site listed below) for descriptions.

Best Regards in Sound & Music Recording,

Leonard Lombardo

<< Subject: Mic for choir - RODE NT1???

From: Markus
Date: Fri, 7 Aug 1998 22:43:05 +0200

Hello!

I want to make harddiskrecordings of our choir (ensemble with about 14 singers, old music), mainly life recordings in church. The mics should be also good for organ. Which microphones can you recommend for me? I want to use two mics for Stereo.
I have a very good offer for a RODE NT1. Does this one fit my needs? What about AKG C1000s in comparison to that? (well, its smaller...) Any other idea in the $300 price range?

Thank You!

>>
Hello Markus,

Your recording situation would be best with a precision matched set of spaced omni microphones. The ambient information captured with spaced omni method also needs to have a baffle between the microphones to make the whole sound coherent during listening playback.

The human head or an accurate acoustic replication is necessary to make the proper working baffle. Any other design goes back to recording sounds in an incoherent manner that the brain doesn't relate to very well and causes much work on the listeners part to imagine the live sound emotion and harmonic relationships.

My web site listed below gives much in-depth background on these issues and contains many reviews of the DSM mics as well as downloadable sounds from of many sound/music sources made with this method and mic.

Many users (see user listings on the Home Page) are capturing the true sounds of acoustic performance in a most easy manner using the DSM method. These recordings are most professional and suitable for later CD replication if desired.

Marrantz has the hard disc recorder that I briefly know about. This has XLR type inputs for mic. The DSM mics can be used with such connectors with appropriate PA-10PFC powering hardware.

A less expensive Sony PCM-M1 DAT deck alternative with equal quality and much easier operation and direct deck upgraded MOD-2 DSM mic powering would seem more efficient and more versatile for easy operation and portability to get to the best recording positions. Digital recordings of value could later be downloaded to a Digital harddisk editing workstation via any portable or standard DAT' deck's digital ports.

Let me know anytime what you think and of any questions.

Best Regards in Sound & Music Recording,

Leonard Lombardo

<< Subj: Re: camcorder microphone question

Date: 9/14/98 8:41:04 AM Pacific Daylight Time
From: Michael
To: GuySonic@aol.com

hey,
thanks for the info and the reference to your comprehensive page, which i visited and will visit some more.
i need to get rid of the hum in the background.
my subjects are documentary interviewees.
they generally sit three to 12 feet from the camera (camcorder).
my problem may be that i can't plug balanced cables into the sony trv9
cheers,
mike
>>
Hello MIke,

Thank you for the reply and the site visit.

The 'HUM" is most likely from the camera motors working against external mic cables with grounding noise problems.

The DSM mics have a 'star quad' mic cable that is quite immune to these effects and will allow the best possible sound recording when used as an external camera microphone.

The Sony recorders mostly feature microphone power at the external mic jack and will power the DSM microphones fairly well without PA adapters.

(see discussion on powering: MOD-2/PA-x Mic Powering Page

Using a PA adapter with Low frequency bass filter will allow reduction of air-conditioning or motor noises from street traffic; reducing this sound can also be done in post production.

For your purposes of interview mic, either DSM-6/H or EH will give good performance and allow use for recording much louder than conversation levels for other ambient sound documentary purposes.

However, for best low noise performance from both the camera and the mic, consider the DSM-1/M a much better interview microphone but, has somewhat limited loudness handling as compared to DSM-6 models.

Mic description on http://www.sonicstudios.com/dsm.htm

Let me know if I can be of service for your requirement or answer further questions.

Best Regards in Sound & Music Recording,
Leonard Lombardo

Subj: Re: [ProAud] Blumlein stereo and Spatial Equalization

Date: 10/25/98
To: pro-audio@pgm.com

This discussion is, in my opinion (and judging from the subject thread length), of vital importance to everyone concerned with the future of true-to-life sound recording. The digital recording era has pressed this subject to new importance as it has the quick ability to expose less than adequate microphones and methods of using these mics.

We must allow that our natural instincts in listening to live verses recorded sound are not being adequately addressed with synthesis of recording techniques within the various methods discussed so far in this thread.

Yes, the method of microphone usage AND playback are very prime in achieving the goal of reconstructing the live event for virtual realistic perception. However, speaker type and placement is second in importance to microphone technique used to make the recording.

Spaced Omni mics (of smallest dimensions and of very careful 'neutral' acoustical design) allow for the most accurate sampling of the sound field. Necessary frequency bandwidth is not an issue with true pressure Omni's.

However, the conditioning of the reception to spaced Omni's is what's important to record all the spatial cues in a humanly coherent manner.

Simply positioning a properly designed HRTF baffle (minus the ears of the binaural method) between the spaced Omni's is all that's necessary to allow for what's now missing or terribly difficult to replicate with the other methods discussed so-far.

I'm sorry that Omni/HRTF method seems too simple and doesn't address the need to be clever with synthesis, but the natural mechanisms of human perception are quite complex and should give us enough satisfaction with just appreciating how well it works when used in full measure within the recording process.

HRTF baffles of correct design are your own head or Sonic Studios DSM-GUY (or to a lesser degree, Lite-GUY). There are no other correctly designed baffles available, period. (See http://www.sonicstudios.com/multitrk.htm)

Omni mics need to be of exceptionally neutral design and quite small to avoid acoustic/mechanical distortions of the sound sampled. There are a few alternatives other than Sonic Studios and Earthworks, but true pressure and ruler flat response from ultrasonic to the subsonic is what's necessary for best results here.

Speaker placement is suggested to be wide and focused to a point just in front of the listener's head. What's important is that the sound field is perceived to be at least 180 degrees with no center deficiency. Careful adjustment of speaker angle achieves this effect and it can hold up for much less than dead-centered listening positions under certain room/speaker-type conditions.

Headphone listening is less satisfying unless the phones drivers are forward of the ears by as much as a few inches and facing back towards the ears. More like the Jecklin Float-Phone Electrostatics, but at a much increased forward projection angle. Sony and others now have phones that attempt this, but the angle is still too shallow. This eliminates the effect of the sound happening more on the sides and around the back with a large hole in the front center.

I know that control over the recording process is paramount to the music at present and that the HRTF method leaves little control over just proper placement within the sound field recorded. However, the benefit is simply recording what we hear in a like manner so that all the gains made with the digital recording technology can be finally fully appreciated without reservation or excuses.

We now have the opportunity to just enjoy recording and listening to fully satisfying and 3-D realistic 2 (or more channel) recorded music/sound without further wanting for much better technology and methods, if only we would allow ourselves to give nature full credit for knowing best.

Best Regards in Sound & Music Recording,
Leonard Lombardo

In a message dated 11/2/98 10:11:12 PM Pacific Standard Time, DAT-Heads-Request@fedney.near.net writes:

<< ---------------------------

From: Len Moskowitz <moskowit@panix.com>

Subject: Re: Re: MIC Cable Suggestions?

Date: Mon, 2 Nov 1998 12:31:02 -0500 (EST)

Len Lombardo <GuySonic@aol.com>wrote:

> Dual Mono XLR-3 female to "Right/Angle molded 3.5mm mini-Stereo Plug" is the
> only safe way to connect standard mics (also Dual 1/4" Female) to Mini-stereo
> jacks without risk of easy damage to the minijack or poor connections.

This is a bit of an exaggeration. Radio Shack makes a nice quality right angle adapter that, in my opinion, is better than using molded plugs. Molded plugs are unrepairable -- a real pain at times.

Len Moskowitz Microphones, Digital Interfaces, Cables
Core Sound http://www.core-sound.com
moskowit@panix.com Tel: 201-801-0812, FAX: 201-801-0912

- >>

I would not recommend using that Radio Shack adapter as it will put exactly the kind of strain on the jack as is suggested to avoid! You end up with a very rigid assembly of connectors that, while at a right angle to the deck, is a rigid assembly when attaching a stereo plug that can just as easily damage a jack with a slight tug.

Don't be foolish enough to take Len's advice on this one. I've known about the Radio Shack Right angle adapter for almost 10 years and would never risk any of my DAT decks with using it.

Sonic Studios Molded adapters have never failed anyone in 8 years of use.

Maybe Len doesn't know how to make reliable ones as yet?

Best Regards in Sound & Music Recording,
Leonard Lombardo

<< -----------------------------

From: Len Moskowitz <moskowit@panix.com>
Subject: Re: Help...what did I do wrong?
Date: Mon, 2 Nov 1998 14:18:55 -0500 (EST)

Eric L <j@netcom.ca> wrote:

> I recently taped a show at mid-size venue, with a D8 using coresound
> binaural mics with roll-off filter. I was set up on a balcony just above
> the soundboard. Instead of clipping the mics to my glasses, I decided to
> hang them over the railing about halfway down. My thinking was that I
> would eliminate chatter from the people next to me and the mics would be
> out of site of security and even if the did see them, they would look
> like strings from my hooded sweatshirt.
>
> The result was my worst recording with coresound mics (although all of
> the rest have been very good-excellent). The sound is not very clear and
> defined (can't clearly hear each instrument) and is very drum heavy.
> Even though the sound from where I was sounded pretty good.
>
> So what did I do wrong and what should I have done?

Seth already covered this but it's worth repeating.

You don't want the mics sitting against a boundary surface, but rather out in free space as much as possible. The reflections from the surface will double the apparent loudness of low frequencies and will create odd sounding phase cancellations in higher frequencies.

If the sound was good where you were sitting, you would have done better by either mounting the mics near your ears or clipping them to an object free out in free space. Spacing them two to three feet apart would have improved the stereo image.

Hope that this helps!
Len Moskowitz Microphones, Digital Interfaces, Cables
Core Sound http://www.core-sound.com
moskowit@panix.com Tel: 201-801-0812, FAX: 201-801-0912
>>

Actually placing a spaced pair of 'high quality matched Omni microphones' against a sold wall or boundary that's facing the sound source is an excellent way to get a 'PZM type' stereo recording that usually sounds very good.

The Crown PZM is based on this principle and many amateur users and a few studio customers have used the DSM microphones in this manner (spaced about 8-24 inches apart against a wall) with reported excellent sounding results.

The pickups must be directly against a hard surface (not inches away). The stereo image is not nearly as nice sounding as with using the headworn method, but it is a good alternative stereo mic technique that gives satisfying results when headwearing the mics is not practical or desired.

Best Regards in Sound & Music Recording,
Leonard Lombardo

<< Subj: HELP!!!

Date: 11/4/98 4:12:29 PM Pacific Standard Time
From: Ph
To: GuySonic

Dear Leonardo

I bought dms microphones 1 month ago, for recording traditional songs in Indonesia.
I 'm leaving for Indonesia on next saturday. I began to prepare my suitcase today and before putting the microphones inside I decide to test them. I'm stupid to not testing them before ! What a mistake because I got problems :
I have a Sony TCD D8
I connect the PA6, with a new battery inside, to the deck in the mic input.
I connect the DMS to the PA6
The deck is connected to a DC in 6V external power
I switch on the deck. Sample 48Khz. Rec mode : manual. AVLS off
Put on my glasses with the DMS attached to them
Sit in front my stereo HIFI, playing the radio, volume : approximatively like somebody speaking loud. Ready. I begin to record.
The level is very low. I switch the mic sens to High. To get 12db on the level peak recorder I have to turn the volume knob to 10 !
I change the PA6's battery for a new one. Still the same.
I stop recording. With my Prodif 24 board and Soundforge I transfert what I've recorded on my computer. I send you a sample.
Scratch, awfull and low sound, nether get something like this, even with a 3$ mike.
Did I make a mistake ? I don't think so as I started the processus again and again, step by step, and each time it's still the same.
When you listen to the recorded sound it seems somebody is touching the plugs. But the deck was on the table, the PA6 too, they were not mooving at all.
So if you are able to help me before I leave... I need the mikes ok for my travel. I'm waiting for your answer. I'm sure you are going to do your best.
Thanks in advance
Philippe
the sample is a wav, about 13s, exactly a copy of what I got on my d8. Sorry it's 1mo sample...

>>
Hello Philippe,

Output level on the H model is a few DB lower than the EH model but will be OK in your H settings if just recording conversation. However, the noise is from poor connections to the deck (and maybe the PA to Mic) that need rotating around or a cleaning; (see mini-connector tips at: http://www.sonicstudios.com/tips.htm/) and look at the noise prevention sheet sent along with your microphones to see how to avoid connection noises.

Ambient sound effects, live music, and street sounds are louder than you might think and the H model should still work well. (DAT is not level sensitive like analog and even lower recorded levels can be normalized much later with no problems with computer processes)

NOTE: DC Power is being sent through the plugs with the mic signal and will show loud noises if not keeping the plug/jacks in best condition.

DO NOT put tape around the plug from the mic and the PA-6 input jack to keep the connection from moving......... this will cause poor mating at this interface. Instead, (if this is a concern), use one of the Velcro ties to capture 'just the two cords' together only.

Rotate the plugs around after cleaning to clear the connection of any residue or until the noises of movement cease......... Rotation noises cease to show a solid, clean connection and is a good way to clear any oxides or contamination (with a little pure alcohol) from the metal surfaces.

DO NOT get any solvent on the mic plug molded plastic.......... it will soften this material and can coat the metal to cause poor connection.... Use a cloth to selectively clean the mic plug metal surfaces if necessary. Both the mic plug and the PA-6 plug/jack have a ProGold conditioner that should provide good lo-noise service if rotated occasionally to remove any noise producing (like fingerprints) material on the mating surfaces.

A little 90+% Alcohol placed on the metal of the plug and rotated in the jacks works very well to get things good for a long time if fingerprint residue has collected inside the jacks (as is likely with the deck's mic input if used a few times before being more knowing about noises and their causes).

Let me know if after doing what's suggested here and on the web site works or doesn't work for you. Problems with noises is common until you follow these procedures carefully at least once.

Best Regards in Sound & Music Recording,
Leonard Lombardo
======================

<< Subject: Recording a Big Band (or any Jazz)

From: "Stefan
Date: Wed, 18 Nov 1998 19:20:18 +0100

Hi!

I play in a Big Band in Norway, and we're going to record a CD this easter.
I've worked as a soundtechnician (as an amateur), but only with live
productions. I've next to no experience with recording.

I would like to get in touch with somebody with some experience with
recording big bands and jazz ensembles, to share some ideas, and maby get a

few tips.

Stefan
>>
Hello Stefan,

A best way to record Big Band is to do an ambient stereo recording.

I and many others have used this approach using my patented method and DSM stereo microphones. See:
http://www.sonicstudios.com/multitrk.htm and http://www.sonicstudios.com/dsm.htm

There's a few sound clips on the site of big band music on the links page: http://www.sonicstudios.com/links.htm

Vocals are more challenging with this method as the PA is usually used for vocals and a special set of speakers place to project directly to the recording position in front of the band is necessary to get good vocals coverage; if included with the band arrangements.

Otherwise, you do the recording within a few meters of the band and adjust to hear a best mix of the entire band. Solos can be enhanced by moving closer to the soloist during those times and backing away to the preferred mix position afterward, all in real time.

Done many Big Band recordings for the local band here and at many festivals over the years, exceeded anything done previous with multi-microphone approaches, spaced arrays, or single stereo type microphones.

BTW, the RAP group is full of studio 'guru' types that seem to have no clue on how to do a really excellent live group performance recording with anything other than using the old and tired mics and use methods that most often don't sound very satisfying, needing a lot of skill and luck to get something that just sounds OK at best.

They seem to wish I would go away so that they could go back to the "old same old" tired way of doing mediocre live recording work without further thought.

Best Regards in Sound & Music Recording,
Leonard & Debbie Lombardo

<< Subj: Re: [ProAud] Are we going about this whole thing wrong?

Date: 1/11/99 8:38:14 AM Pacific Standard Time
From: bobkatz@digido.com (Bob Katz)
Sender: owner-pro-audio@pgm.com
Reply-to: pro-audio@pgm.com
To: pro-audio@pgm.com

><<What about a light coat of paint instead, on top of one generation of 48
>bit-processing plus 24 bit dither?>>
>
>Mmmmmm, shifting sands, Bob! (as you don't need me to tell you, I dare

I'm shifting sands with questions, lots of questions, Rich. The basic question is: If a noisy analog tape sounds so good with -60 dBFS of noise (or worse), why can't we find a solution to 24-bit digital recording that masks the last residual high harmonic distortion? Are we looking in the wrong place by trying to get rid of distortion we can't possibly get rid of? If the last distortion that's left is the ugly kind, why not add just a touch of the "good" kind to mask it?

All I'm doing so far is asking questions....I'm not abandoning my search for purity by brute force exercise of high precision....but it's just sure strange when you think of it that the more we work in high resolution recording, the more ugliness is exposed instead of made hidden.

BK

>>

Great thread!! This is my all time favorite subject and one that is welcome to see finally discussed here.

The Media in which music is mixed is (or was) air; the acoustic 'in the ambient' mix is perhaps considered to be the 'perfect mix' of all the elements that make up the composition. It's the most musical when just 'heard', as the live sound experience relates well to this process.

When sound is gathered from all non-acoustic mixed sources (multi-mic and/or pure electronic to analog or digital domain mix mechanisms) the 'quality factor' of the mix is heard as much less than with the 'perfect' acoustic mix.

Analog mix boards often use capacitors (a form of transducer) and bandwidth limited stages to sum the signals from non-hamonically related sources that more often sounds better than the digital domain processes that just work in a mathematical domain.

Mixing disassociated harmonic sounds to analog tape may be thought to be performing a 'healing or blending' of non-related harmonic content by functions native in the analog deck's electromagnetic processes. Certainly the addition of 'musical' harmonic distortion is a factor here to be considered as very important to the 'healing' of a mix of non-harmonically related sources.

I've personally noticed a lack of this problem when the recording is sourced from a more natural microphone method where most all the harmonics in the acoustic air 'perfect' mix are captured in a psycho-acoustically correct and coherent manner. Recordings made with this method of acoustic microphone method show the true worth of digital recording in any bit/resolution depth.

While acoustically outputting all tracks of unrelated sound sources and re-recording all at once with a suitable (stereo or multitrack) microphone method that captures this acoustic mix in a coherent manner certainly works very well, it's not (yet) considered very practical for most modern projects. Mix to analog tape is a bit more practical, but also is not as effective as the acoustic mix process and is using electromagnetic technology that's one-two steps away from the scrap pile.

Replicating what happens acoustically in the 'perfect' air media mix in a digital process seems the 'holy grail' of this subject. One that may take even more powerful processing than is to be believed. Using a natural microphone method as often as possible to record those acoustic instruments and performances is another practical technique until a 'natural' digital mix process is better understood.

Using advanced analog mixers with non-coherent source 'harmonic blend/enhance' ability may be the most practical solution until digital processing is at least an order of magnitude more efficient and/or faster.

Harsh Digital sound is much more a problem with non-coherent harmonics from multitrack methods of sound compilation than a fault with digital recording itself. The use of analog mixers and tape only allowed some rather poor and little understood recording methods/practices to be passable. Digital processes are not nearly so accommodating as analog in piecing together non-acoustically related recorded tracks.

Recordings made with natural ambient mic methods are the best available for getting the sound right from the very start. But this is not what most multitrack addicted recordists want to hear as this means a tremendous loss of 'control' and surrendering to accepting of what-you-hear, is what-you-get recordings. However, there is bound to be an audience of some size who would appreciate more recordings made in this manner of all kinds of music styles.

Anyway, it's just hard as hell to fool with Mother Nature in the music-harmonics domain.

Best Regards in Sound & Music Recording,
Leonard Lombardo

<< Subj: microphone question

Date: 1/11/99 10:47:42 PM Pacific Standard Time
From: charles
Reply-to: .com
To: GuySonic@aol.com

I wonder if there's a way now known or a microphone that has been developed that would enable me to perform voice-over recording without extensive sound-proofing in my home studio.
I currently record professionally, with a great deal of success, from my home but even with a closet framed out to be a "sound-proof booth", I still must be careful to turn off the central air conditioning unit and hope no plane flies over while recording.

Before investing in an expensive acoustical booth, I want to continue to research the full range of microphones which may be available that would be directional enough to record the best aspects of my voice while blocking out other noises and annoying frequencies, etc.

Any suggestions?
--
- Charlie

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Next day voice-overs on DAT
>>

Hello Charlie,

Constant and steady background noise recorded into the microphone you prefer for quality-of-vocal reasons can be greatly reduced (to a varying degree) using noise reduction features of software like Cool Edit. Other less constant background noise is not (easily) removed by this method.

Sitting in a well constructed 'luxury' automobile parked in an insulated (walls/ceiling) closed garage is quite effective as an isolation booth and may be (partially) tax deductible for your purpose!

Best Regards in Sound & Music Recording,
Leonard (& Debbie) Lombardo

Subject: Re: How to stereo mic Grand Piano?

From: Eleven Shadows <elevenshadows@santa-monica-ca.com>
Date: Fri, 05 Feb 1999 01:20:56 -0800

Michael Vladimirsky wrote:

> Sorry I do not know the AudiTechnica mics you mention. If I were you I'd
> borrow a second C1000 and record the Yamaha with the lid open in close AB,
> from 1.5 m. distance. The mics ahoudn't be parallel. Adjust the angle to get
> the stereospread you like.
>

I think the ATM 31s are small diaphragm condensers that are similar to ATM 33Rs and ATM 33a mics, but someone correct me if I'm wrong. If that is the case, you can record the Yamaha with the lid open. There are a zillion ways to record a piano, but what I would do is try x-y over the piano, which usually works. I don't know what kind of music you are doing, but as a general rule of thumb, for more New Agey sort of stuff, get them farther away from the hammers; for certain kinds of classical and rock, get 'em a little closer. However, with a lot of classical, having those mics backed off and to the right of the piano player gives a more natural sound. Use your ear. A lot of people do spaced pairs, choosing to hover the mics over the strings that give it a more dynamic stereo spread.

What I personally like doing is opening the lid, but having the mics backed off so that they are outside, and placing the mics at least three feet from the instrument. To my ear, this allows the sound to blend and sound more natural. If you have a really great sounding room, try a spaced pair of omnis. If you don't have access to that, you can try the two Audio Technicas. I like to try both spaced and x-y and see which each result gets.

If it's rock piano, it's often heavily compressed to blend in with the rest of the music. In either case, the Audio Technicas are probably pretty bright mics, and I personally find that bright mics frequently help with the piano sound quite a bit.

> Michael Vladimirsky

> russian professional microphones
> +7(095)1906152
>
> <19990205014812.26737.00000949@ng-ca1.aol.com> ...
> >I am going to record a Yamaha grand, 6 to 7 foot, in stereo. I have SM57, C1000, two ATM31s, an old small D Audio technica omni, and an Astatic mic that is a lot like a SM57. Going through a couple single channel Tube MPs into an Akia DSP12 Hard disk recorder. Any suggestions on mic choice (other than get better ones) and set up?
> >
> >Craig Birchfield
> >
> >"You can check your anatomy all you want...you get right down to it this far inside the head it all looks the same---No, No, No, don't tug on that! You never know what it might be attached to." >Buckaroo Banzai (while doing brain surgery)
> >

--
Ken/Eleven Shadows, looking for a Super 8 camera and Super 8 projector
~~~~~~
Eleven Shadows * ES songs on Real Audio * Music Reviews * Travels:
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~~~~~~>>

As Ken stated "there's a zillion ways to mic a piano" for recording purposes.

But there are just a few less than a 'zillion' ways to get an acceptable 'piano sound'. There are even fewer ways to mic for 'THE sound of a piano'.

Placing mics using the 'here and there' way, will reliable get that 'piano sound' we've grown accustomed to hearing in popular music tracks and there's almost a 'zillion' ways to approach this to get a unique 'piano sound' to fit the any occasion.

However, If wanting to mic for a realistic 'sound of a piano', the 'here and there' way of mic placement is NOT the way to go, but ANY of the stereo microphone methods discussed so far IS A WAY for intention of recording a convincing 'real piano' sound. To succeed with this however, presents far fewer ways (choices) of mics and 'using method' and presents a far greater 'skill level' challenge and/or being 'very lucky' to consistently get satisfactory results. Far from being impossible, it's at least much more difficult for a number of good reasons.

The intention of recording piano realistically involves a stereo mic method as the output of these is two channels much like our own two channel hearing way of hearing sounds. So stereo mics need to record two different perspective that will most satisfy our natural hearing sense.

For us to be convinced of hearing a real piano within a recording, the stereo mic must record sound in a 'unique' way that includes 'psycho-acoustical' information within the two tracks of recorded audio. While the 'psych-acoustical' information necessary for us to hear a convincing sound of a piano is 'unique', the uniqueness of the stereo mic/method of using such, should not be TOO uniquely different from our own way of 'uniquely' hearing sounds.

And here's the rub of the stereo mic methods discussed so far: They can easily be far TOO unique and present only a 'partial set' of proper psycho-acoustical cues; often including (free of charge) a whole new unique set of strange (to our normal hearing) sound cues that are not 'coherent' or recognized (without doing 'mental' conversion type interpretation work) as part of a real sounding piano.

The stereo mics discussed so far are unique to each other (including us) in larger or smaller degree with mic placement rather critical to each new ambient situation. Because critical placement is often different with each 'type' of stereo mic/method (assuming the same ambient working condition), being able to listen yourself for an acceptable 'heard acoustic mix' of instrument and ambient (room, hall, etc.) is ALL IMPORTANT.

However, because of the degree of 'TOO much uniqueness' of each stereo method discussed, just listening will not reliably work unless you're (as mentioned earlier) very lucky. What you hear is NOT OFTEN ENOUGH what you'll record with stereo microphones and you'll need a lot of experience, luck, and/or time for the 'trial and error' record/playback procedure necessary to avoid disappointment from having assumed too much.

I would be much nicer to learn to quickly hear a microphone position (music + ambient mix), plunk the microphone right there, and roll tape (or spin hard drive) and be much more assured of getting what you heard because the stereo microphone is not so unique to our own perceptions of sound.

There's only one stereo microphone 'way' that'll consistently allow the 'what you hear is what you record' assumption regardless of situation. That microphone is a 'Head Related Transfer Function' (HRTF) type of stereo microphone that uses a unique baffle between two very small, precision matched omni mics.

This type of stereo microphone is rarely discussed or mentioned (at least here) perhaps because it's TOO much a 'no brain'R'??

Not being challenging, needing much skill, being lucky, or having the immense joy of doing multiple tests/retakes makes this type of stereo microphone hard to act expert about for sure, and may as such, be generally ignored by the standard knowledge base of available microphone experts.

As far as I can tell from being around here for over 5 years, it might just be working too well(!) dampening the joy of endless discussion of all the challenging ways 'uniqueness' in microphone 'perception' adds to our pursuit for convincingly real (ambient acoustic) recordings (if that's your aim).

If mic/method solves a lot of previous problems, what will the 'problem solvers' now do? This remains a real 'bureaucratic type' challenge and seems worth much discussion of what to do next when 'favorite' discussed problems are threatened to be solved for good, making other options less accessible in appearing like good advice.

Fortunately, I'm here and again helping those who truly desire to make their recording more consistently real acoustically sounding with the experience, good advice, and the hardware to back up where my mouth is. As with many expert recordists, microphone companies refuse to adopt new microphone designs while the inventor still lives and breaths. If you doubt this, look for finding the persons responsible for the classic stereo microphone methods discussed here, (Blumlien, Soundfield, etc.) they've virtually all died years before any of these 'now highly discussed' methods were allowed real commercial production/availability or regarded as a mic technique worthy of discussion.

Things being as they are, no need to wait till I'm 'dead and gone' to get THE stereo microphone right now (only lacking any discussion of such from those 'teaching' the old standards of recording art), as I'm one of those very rare inventors that is able to produce products without the 'recording industry acceptance' due or scheduled sometime after my passing.

My web site has the necessary details on THE stereo microphone that is very NON-assuming or in most ways non-unique to how we hear sounds; what your hear is exactly what you'll record; 'relearning' to trust normal hearing IS going to be tough on the 'old timers' used to sticking one finger in an ear between retakes. But while there's still life, learning is possible!

Please, don't all thank me at once for my dedicated efforts! Just go out there and make it sound more real for the old GUY!

Best Regards in Sound & Music, Leonard Lombardo
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"A bit of knowledge coupled to a great deal of wisdom serves us best"

Subject: Re: How to stereo mic Grand Piano?

From: guysonic@aol.com (GuySonic)
Date: 08 Feb 1999 00:43:40 GMT

In article <36BC08EC.86AD1A53@santa-monica-ca.com>, Eleven Shadows
<elevenshadows@santa-monica-ca.com> writes:

>Subject: Re: How to stereo mic Grand Piano?
>From: Eleven Shadows <elevenshadows@santa-monica-ca.com>
>Date: Sat, 06 Feb 1999 01:18:38 -0800
>
>> This type of stereo microphone is rarely discussed or mentioned (at least here)
>> perhaps because it's TOO much a 'no brain'R'??
>
>The question was "how do I use ***the mics I have*** to get the best sound."
>This is the original question:
>
>> I have SM57,
>>C1000, two ATM31s, an old small D Audio technica omni, and an Astatic mic
>that is a lot like a SM57. Going through a couple single channel Tube MPs into
>an Akia DSP12 Hard disk recorder. Any suggestions on mic choice (other than
>get better ones) and set up?
>
>Read the last sentence again. Great. Now read one more time and try and understand what the guy is saying.

>
>> Fortunately, I'm here and again helping those who truly desire to make their
>> recording more consistently real acoustically sounding with the experience,
>> good advice, and the hardware to back up where my mouth is.
>>
>> My web site has the necessary details on THE stereo microphone that is very
>> NON-assuming or in most ways non-unique to how we hear sounds; what your
>hear is exactly what you'll record; 'relearning' to trust normal hearing IS
>going to be tough on the 'old timers' used to sticking one finger in an ear between
>> retakes. But while there's still life, learning is possible!
>>
>> Please, don't all thank me at once for my dedicated efforts!
>
>If you're that hot of an engineer, you should be able to get a good sound
>with inexpensive mics. Put your money where your mouth is and tell this guy how
>to get a good sound with the equipment that he has. I've tried to help him. Why
>don't you?
>
>--
>Ken/Eleven Shadows, looking for a Super 8 camera and Super 8 projector
>~~~~~~
>Eleven Shadows * ES songs on Real Audio * Music Reviews * Travels:
>Peru-Ladakh-Kashmir-India-HK * Tibet * Real Audio Radio Shows
>http://www.theeleventhhour.com/elevenshadows
>~~~~~~
>

Ken, you're correct in pointing out not regarding his equipment limiting clauses enough and I was going mostly on the thread discussion that followed + the original post's stated desire to record a "piano in stereo". I hope the possible (over) dramatic portions of my past reply didn't completely mask the useful concepts presented.

> If you have a really great sounding room, try a spaced pair of omnis. If you
> don't have access to that, you can try the two Audio Technicas. I like to
try
> both spaced and x-y and see which each result gets.
> ------------ (Ken's post)

>A piano is a mono source? I mic grands from 10-15 feet away using a Blumlein array and even at that distance, the piano isn't mono. The room reverb isn't mono either but that is another discussion.

With piano, there can be a lot of experimentation. the sound can be a distant room sound all the way up to the very close "head in the lid sound" and this is just from a Blumlein array.

Other techniques such as miking above the pianist's head or from the far end of the piano will produce results that may be pleasing for different styles of music.

AB, XY, MS, ORTF, Blumlein, and other techniques will give different results. Some good some not-so-good.
-
Richard Kuschel
>

If he does have a pair of resonably matched omni (at least two of the same model/production run), then placing these two mics on a HRTF (like the LiteGUY) baffle will allow an easier way to record 'the sound of a piano" with minimum expense and skill level. If he doesn't have or want to get those two omni's you mentioned, then 'he can't get there from here' using any other type of mic if recording a realistic sounding piano is the goal with the available mic limitation imposed.

Then Richard opened the 'playing field' with naming (most all other) stereo microphone methods regardless of mic closet limitations. So here we are again.

Ambient stereo recording of convincing quality involves correctly matching the perspectives of the microphone method with our own as accurately as is possible. The stereo microphone (mics/methods) previously mentioned, fall far short of this task and don't look at all like 'us' in capturing ambient acoustic information, but provide plenty of recorded variations that are 'interesting' if not occasionally quite satisfying.

Our hearing reception is neither directional like the Cardioid nor like spaced microphones floating in space. Our baffles (our head) is not a reflective disk (like with the Jecklin Disc) nor made of plastic or foam (like the Achen and Nueman baffle). In short, all these approaches remain as using methods and materials that have far too little or no 'natural' acoustic perception mechanisms along with a whole lot of 'unnatural mechanisms' by design. How a recording engineer gets the best stereo performance from such mic systems remains the true 'art' that's most discussed. Unless you really need to reject some portion of the ambient (suffering the usual image/sound distortions) considered as interfering 'noise', there is only disadvantage to using direction microphones in any kind of simple or fancy (matriced) array.

The acoustic recording process is only complex because acoustic sound is complex and our recognition of percieved sound is also complex. However, if the (stereo) microphone's processing method is kept eloquent and natural (to us, like HRTF baffled omni pair), then just a simple microphone system is all that's needed to report the complexity of the acoustic ambient to our also complex recognition mechanism via recorded medium.
This is more a situation where the preferred microphone/method does not 'get in the way' of making an accurate (to our perspectives) record of ambient sounds or does not add unneeded confusion with strange ways (to us) of acoustic/electrical signal processing.

Your choice of a suitable omni mic pair to use with HRTF baffle is really up to you as there are plenty of candidates available, some with less than flat response that might be exploited for providing brightness or similar tonal effects. For the most accurate tonality/image, small sized capsules of extended ruler flat hi-frequency bandwidth have a decided advantage. Choices of available HRTF baffles remains much more limited to those shown on my site.

Of equal importance to piano recording is using stereo method recording for other acoustic instruments from flute to drum kit when depth and realism (as you'd hear it) sound is the intention. With this, the ambient you're working in can be a nightmare jungle to a paradise of sound (fill) quality. Using acoustic traps (like ASC's) can help take most of the jungle out of your available working spaces; giving much control of the ambient quality and allowing good results with trying less-than-close mic'd (mono or stereo) ambient recordings in general.

On the Super 8 equipment, many of the camera repair shops (especially in the larger metro areas) used to keep quite a few of these around for spare parts if for nothing else. Partially reconditioned with some cleaning and new lubrication (very important), these should be available for just slightly more than a song these days.

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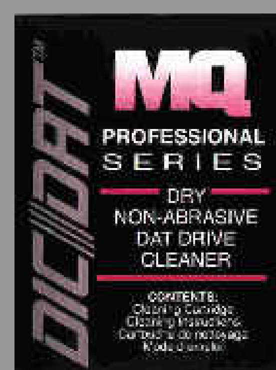

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