TASCAM US-122 Review

A shorter version of this review appears in Recording Magazine, December 2003

TASCAM, with partners Frontier Design Group, jumped on to the USB (Universal Serial Bus) audio hardware bandwagon early, turning out a broad line of products over the past three years, the US-122 being the latest. While other members of this family offer a recorder- and mixer-like control surface for the DAW (digital audio workstation) software in addition to providing multiple audio input and output channels, the US-122 is a departure. It's a two-channel 24-bit, 44.1/48 kHz audio interface, leaving DAW control to the traditional mouse and keyboard. Much more than a USB "sound card," it solves many real world problems encountered by nearly every 2-channel sound card, neatly integrating functions typically performed by the very outboard units that the DAW user hoped to eliminate.

What's Its Schtick?

In a nutshell, The US-122 provides extensive control over monitoring during both tracking and mixing. The two biggest problems it solves is hearing yourself in the headphones when overdubbing without having to fuss with tiny buttons on the DAW screen (or worse, a driver or mixer applet), and providing real knobs for independent volume control of speakers and headphones. Throw in a pair of serviceable microphone preamps with switchable 48 volt phantom power, DI inputs for instrument pickups, and insert send/return jacks for patching an outboard compressor or equalizer into the recording path, and you have a very comprehensive input and control system for a tabletop or portable recording studio.

The US-122 also sports a single port MIDI interface and it comes bundled with Steinberg's Cubasis multitrack DAW program (Windows and Macintosh pre-OSX) plus TASCAM's own Gigastudio 24 (Windows only).

The Basics

The US-122 is a tabletop box, a tad larger in the flesh than I expected from the photographs, but logically laid out and heavy enough so that it won't accidentally get knocked off the table. Knobs are comfortably sized and well spaced, switches and connectors are well marked. Indicators are provided for phantom power, input signal present and near-overload, MIDI activity, and USB communication. Input and insert jacks are located on the front, USB, MIDI, line output, and headphone jacks are on the back. Power is supplied through the USB port so there's no wall wart, batteries, or power cord. To keep cabling neat, I'd prefer to have the headphone jack on the front and the inserts on the rear, but the real estate just isn't there. Inputs and outputs, with the exception of the computer connection of course, are strictly analog – no AES/EBU, S/PDIF, or optical I/O.

Mic Inputs are balanced XLR's, line outputs are RCA jacks. All the other jacks are full sized (1/4") TRS connectors, with the shared Line/Guitar Input jack offering a balanced input which works fine with unbalanced sources too.

While the overall feel of the unit is robust – no rattles, and a good heft - the knobs feel a bit shaky. The guts of the USB-122 are on a single horizontally-mounted circuit board with the pots secured only by their circuit board solder tabs. There's some mechanical support for the pot shafts molded into the plastic top cover, but still the knobs wobble a bit. I doubt that there's risk of breaking the pot shafts or soldered connections in normal use, but if you're planning a mobile lifestyle for your US-122, take care when shoving it into that already overstuffed gear bag.

Phone jacks, too, are secured only by their circuit board mounts. Like the knobs, they're constrained from sideways movement by the case molding, but they wiggle if you don't pull a plug straight out. This is a potential point of mechanical failure if you're careless, but hey, what can I say? Fifty years of using gear with pots and jacks secured to a panel with nuts have just made me expect a really solid feel. On the positive side, the 1/4" jacks grip the plugs solidly, so I'm not concerned about losing a connection unexpectedly. The XLRs are typical of those found on small mixers – solidly mounted but with no latch, just a friction fit to retain the mic cables.

The only accessory packed with the US-122 is a shielded 4-1/2 foot long USB cable. TASCAM suggests that you not to substitute another cable, probably because it's shielded and they've tested it for maximum throughput. While I appreciate the full sized headphone jack, including a mini phone plug adapter for the benefit of those who have that style plug on their phones might have been a nice gesture.

Gozintas, Gozoutas, and the Scoop on Specs

Inputs are through an XLR or 1/4" phone jack as appropriate for the source. Each of the two input channels has its own Mic-Line/Guitar switch, so you can mix sources, for example recording a mic on one channel and a guitar pickup on the other. There are two LEDs associated with each input indicating signal presence (SIG) and overload (OL). These monitor the signal level after the input level control and insert jack so they can keep you out of danger if you goose up a particular frequency range too much with a patched-in equalizer. The OL indicator lights when the digital level reaches a bit less than 2 dB below full scale. There's 35 dB between where the SIG and the OL LEDs light, a reasonable guide to a workable record level during setup, but keep an eye on your software's meters when recording.

At maximum gain, the microphone input requires –74 dBu to turn on the Signal LED. There's enough gain to reach full scale on peaks using most modern

popular dynamic and condenser mics on a reasonable source at a reasonable distance, but it's a little shy for some ribbon microphones. At maximum gain, I had to bellow pretty loud into a Beyer M260 (ribbon) in order to tickle the OL indicator, but a Studio Projects B1 (one of those amazing $99 condenser mics) could get me to full scale from my singing voice with the level control at a comfortable 10 o'clock position. With the input level set to minimum, the Clip LED lights at –2 dBu, allowing you to get pretty close to a loud source without overload. The mic input is transformerless, with its input impedance around 2 kΩ, a reasonable match for today's popular condenser mics.

Editorializing a bit here, I'm continually dismayed at how many people believe that a preamp is deficient simply because they have to turn the gain up all the way to get a good recording level. Don't sweat it, and crank it if you need to! The noise out of the US-122's preamp increases about 2.5 dB when going from minimum to maximum gain. Adding a little extra noise at the input is better in the long run than recording at too low a level and having to amplify afterward in order to achieve a competitive playback volume. Why? One reason (though somewhat insignificant with 24-bit recording unless you get absurdly low) is that you gain resolution by using more of the available dynamic range. A more practical reason is that you can hear what's going on noisewise as you track and won't surprised by a noisy track at mixdown time.

A single phantom power switch engages 48 V power on both microphone inputs. There's plenty of juice, enough to provide in-tolerance voltage to two microphones drawing 10 mA each. There are, however, a couple of mic powering caveats that I'll discuss later. Stay tuned.

Line inputs are balanced, but they work equally well with unbalanced sources. At minimum gain, the Clip LEDs come on at +18 dBu. This will accommodate a nominal +4 dBu output device, allowing 16 dB of headroom before digital clipping. With a –10 dBV source, there's enough range on the control to allow as much (or as little) headroom as you choose. The XLR and 1/4" inputs jacks are essentially tied together – plugging into the line input jack seriously loads the mic and vice versa, so you can really have only one input source connected to a channel at a time.

The analog input stage has enough, but just barely enough headroom – it starts clipping about 1 dB beyond full scale digital level. You won't see that clipping in your recordings because you'll reach digital clipping before it happens. Analog distortion, however, while acceptably low, starts to rise about 2 dB before the OL LED illuminates.

The insert jacks, both send and return, have a maximum operating level of +6 dBu. With the input just below clipping, the Insert Send output level is +6 dBu, and an Insert Return level greater than +6 dBu will drive the A/D converter to clipping. This makes the Insert a good match for devices with a nominal –10 dBV

operating level and it should work well with most stomp boxes. If you're using a nominal +4 dBu device in the insert path, although it will work just fine, you won't get a lot of action on the processor's meter and may compromise it's signal-to-noise ratio. Furthermore, if you're using a processor with a calibrated threshold control such as a gate or compressor, you'll need to mentally re-adjust your concept of "0 dB."

The gain of the Guitar input jack falls between that of the mic and line inputs, clipping at –24 dBu when wide open, and accepting +12 dBu before clipping at minimum gain. I was able to drive it into overload with a guitar, but you don't want to do that because it sounds ugly. Get your creamy toob distortion from a processor ahead of the input, a stomp box in the send/return loop, or from a plug-in when mixing.

The manual specifies the impedance of the guitar input as 910 KΩ, but it isn't, at least not on the unit I had for test. With the Guitar input selected I, measured about 15 KΩ impedance at the input jack, practically the same as the line input when fed from an unbalanced source. This explains why the guitar I used for testing sounded a little duller when plugged into the US-122 than with a good DI or another preamp with an instrument input. When I asked my buddy at TASCAM about this, he said "I'll ask the people in Engineering" (which I assume are at Frontier Design) but he never got back to me with a reply before deadline. I suspect an error in the circuit board layout, but without a schematic I couldn't confirm it. Hopefully this will be corrected in later production and can be fixed on existing units.

Line Outputs are unbalanced, -10 dBV nominal operating level on RCA jacks. With the Line Out control fully clockwise, a 0 dBFS recording plays back at +6 dBu. The line output can push +9 dBu before clipping, and there's good reason for that 3 dB of headroom – it allows room for mixing the input source with the playback without clipping the output. That's an important function which I'll get into a bit later. Since you'll probably mix in the computer rather than to an outboard recorder, the Line Output is a good place to connect your control room monitor speakers. You'll need powered monitors or a power amplifier that will give you ample volume from a nominal -10 dBV source.

I found the headphone output, even when fully cranked, to be a bit wimpy. It's difficult to measure acoustic headphone output accurately without the proper test fixtures so you'll have to trust my ears on this. I attempted to make some rough comparative measurements using my trusty Radio Shack sound level meter held up to an earphone while playing music, tones, and pink noise, but got inconsistent results. Suffice it to say that for the same test recording, the US-122's headphone output was noticeably lower than the headphone output of my Mackie 1402 VLZ-Pro mixer, my Soundcraft 600 console, or even my TASCAM CD burner. If you're the drummer, you might need to use an outboard

headphone amplifier, but an acoustic player or singer should get along fine with the built-in headphone amp.

Noise and Distortion

With the monitor gain set for a reasonable playback level in a quiet room, I could hear a faint but continuous popping noise coming from the US-122's line output with no input. The noise was independent of the setting of the Line Output control, so it's coming from something beyond that control, possibly a noisy output capacitor. I'm convinced that it wasn't RFI picked up by in the cables to the monitor amplifier since it went away when I unplugged the USB cable. Broadband noise at the Line Output measured around -81 dBu, or –90 dBu A-weighted (TASCAM's spec sheet uses A-weighting for their noise measurements, which isn't much like real life but it makes the numbers look better). They clearly specify the noise level right at the output but I believe it's important since it defines the noise floor of your monitoring system or, more important, your mix if you're mixing to an outboard recorder.

Looking at the quiescent line output with a 1/3 octave spectrum analyzer, there's a peak at the AC line frequency of about –86 dBu, and otherwise, white noise. The presence of 60 Hz, coupled with the observation above that output noise is independent of the setting of the level control and disappeared when disconnecting the US-122 from the computer suggests a ground loop in the output path of my test setup. As difficult as it can be to eliminate ground loops when a computer is involved, I'm willing to let this pass. Bottom line – If you crank the listening volume up too loud, you'll hear a some noise, however it shouldn't be a real problem at any reasonable playback volume. The good news is that these pops don't appear on the recording, so when a client asks "What's that noise?" you can truly say "It's only on the monitor."

To check noise performance of the inputs, I made a test recording with the microphone input terminated by a 150Ω resistor (my "dummy mic") and the gain set to minimum. In practical theory (if I may be permitted this poetic injustice), this is as quiet as a microphone recording can get. The resulting recording consisted of white noise peaking at about -82 dBFS. This is the noise floor for recording.

Total harmonic distortion plus noise (THD+N) from mic input to playback output is less than the 0.02% THD of my generator so I couldn't measure it directly. Playback of a –3 dBFS computer generated tone (which presumably has the lowest possible distortion) yields an acceptable, but not outstanding 0.02% THD+N. For comparison, my Digigram VX Pocket card in the same laptop computer that I used for testing the US-122 measures –92 dB broadband quiescent noise and 0.005% THD+N when playing the same –3 dBFS computer generated file. This suggests that the THD+N which I measured in the US-122 is

largely due to the quiescent noise of the output stage, and its recording performance is actually a little better. That's a good thing.

Direct Analog Monitoring

Here's the feature that distinguishes the US-122 from other computer audio interfaces. With a typical sound card based recording setup, there's always some delay between the time the signal enters the mic and when it comes out the headphones. The signal zips through the analog input circuitry, then meanders through the A/D converter, to the sound card driver, into the computer where it's routed back through the driver again, through the D/A converter and finally out to the headphone jack. This takes some time – the delay between input and output is rarely less than 1.2 milliseconds, and people have reported tens of milliseconds.

A long delay is perceived as an echo. This can be distracting, but even more annoying for many (including me – it's one of my pet peeves about computer recording) is what happens when the sound from the headphones arrives at your ear just slightly later than the direct sound through the path in your head. Your ear mixes those two signals, producing a comb-filtered version of your voice with dips at frequencies which are multiples of the inverse of the delay time. Short version: Your voice doesn't sound right in the phones.

This comb-filtering isn't present during playback since there's no direct vocal sound involved, but it can be distracting when you're focused on controlling your voice. Obviously if you try to adjust EQ while listening to a comb-filtered version of your own voice in the headphones, you're sunk. One solution which I'm afraid is all too common is simply to turn the headphone volume up loud enough so that it swamps out the acoustic sound. This makes the ratio of the amplitudes of the two signals large enough so that comb filtering is negligible.

A more satisfactory (but more gear-intensive) solution is to use a small mixer as a front end to the sound card. By feeding the headphone amplifier from a bus on the mixer, you can bypass the round trip through the computer. By bringing the playback of the previously recorded tracks into another pair of mixer channels, you can balance levels between the playback and what you're recording in the current pass.

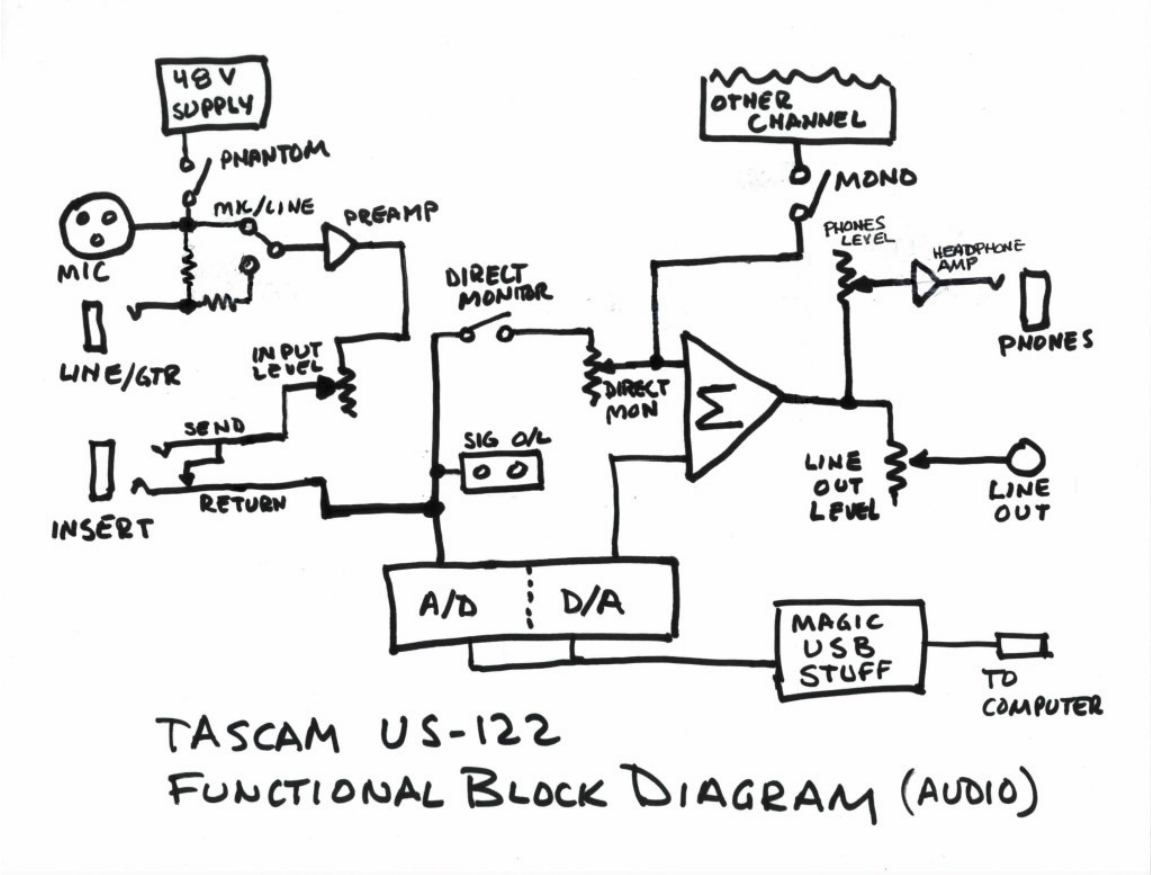
The US-122 implements this auxiliary mixer through its Direct Monitor switch and associated level control. Switching on the Direct Monitor taps the input source and sends it, mixed with the playback off the computer, directly to the line and headphone outputs without the trip through the computer. The Direct Monitor level control allows you to control the balance between the "live" input and playback. Once you learn how to switch off the input monitor function of your DAW software (Cubasis, supplied with the US-122, has a button for this) direct

monitoring works really slick. No echo, no comb filtering, and one-knob control over how loud you hear yourself in relation to the other tracks while overdubbing.

Curiously, the US-122 manual states: “[the buffer size adjustment] does not affect the latency of the US-122’s hardware input monitor, which is always less than 1.5 ms.” A lot of things get called “latency,” so they may be talking about something else, but the US-122’s direct monitoring has, as expected, no measurable delay from input to line or headphone output. Clearly it takes no detours through the A/D and D/A converters or the driver.

Signal Flow

Let’s take a tour of the block diagram and see how this all works together. Note: TASCAM was unable to provide me with an official block diagram, so I drew this one out myself based on signal flow through the US-122. It’s not guaranteed to be completely precise but it’s how things work.



Input starts at either the XLR or 1/4” phone jack. The input signal is amplified, then goes through the Input Level control to the Insert jack (so, yes, the insert here is post-“fader”). Note that there’s no attenuation ahead of the preamp other than the Mic-Line/Guitar selection so it’s possible to clip the input stage with a really hot signal. I didn’t encounter this in practice however, and it’s doubtful that you will either.

The Insert jack is normalled for uninterrupted signal flow unless a plug is inserted and an external device is patched in line. The signal splits into two paths at the insert return point. One path goes to the analog-to-digital converter (the SIG and OL indicators monitor the level at this point), the USB port, and on to the computer. The other path is for direct monitoring - with the Direct Monitor switch on, the input signal goes through the Direct Monitor level control and into the internal mixer.

The digital-to-analog (D/A) converter processes data coming back from the computer, and its output goes to the internal mixer. The mixer output branches in two directions. One path is to the Line Output jacks through the Line Out level control. The other path is to the Phones jack through the Phones level control and headphone amplifier. The Mono switch parallels the direct monitor path of the two channels, placing the input to a single channel in the center of the monitor mix when using the direct monitor function. It does not sum the channels of the playback off disk - that's always stereo and panned by the controls in the DAW's mixer.

For the specheads, A/D and D/A chores are handled by a single AKM Semiconductor 4552 VT CODEC chip. Analog chores are handled by little itty bitty surface mount chips too small for my old tired eyeballs to identify.

The Computer Connection

Computer hookup is simplicity in itself, just plug in the USB cable, install the drivers, and you're ready to play. Drivers are supplied for both Windows and Macintosh computers, however the only Mac I have here doesn't have a USB port, so all of my testing was done under Windows XP and Windows 2000. Both use the same driver set, and "set" it indeed is. There are three drivers (ASIO, WDM, and MIDI) to load. During initial installation, it appears like you're going through the installation process three times but you only need to do this once - almost. The drivers must be installed with the US-122 connected to a USB port, and if you later connect it to a different USB port on the same computer, Windows sees it as new hardware and will politely ask if you want to install the drivers. At that point, you go through the "three times" routine again, though there's no need to use the installation CD since the drivers are already on the computer's disk and Windows finds them. Having done that, you can plug it into either port and the US-122 will be recognized immediately.

The installation CD also contains a copy of Cubasis. I installed it, verified that it works fine with the US-122, and after concluding that I simply didn't like the user interface, set it aside in favor of some more familiar software. A second CD installs Giga drivers and Gigistudio 24 which plays fine (to the very limited extent of my playing) with a keyboard plugged into the MIDI IN port.

An interesting feature of the US-122's drivers (and in fact all of the Frontier Design drivers for this series of TASCAM products) is the ability to mix the outputs of both the ASIO and Giga drivers in the same application without resorting to outboard programs such as Rewire. When using a program such as Cubasis, you can simultaneously play and mix both the recorded audio virtual Giga instruments as if you had two separate sound cards even though they're handled by the same hardware.

The small instruction manual walks you through a simple Cubasis tracking and overdubbing session, but you'll have to rely on the on-disk manuals for Gigastudio and more details on using Cubasis.

Some tweaking of the computer's operating system and driver settings is almost always necessary in order to get an audio interface to work smoothly without stuttering, and the US-122 is no exception. The instruction manual includes a number of tips, the simplest being to disconnect any unnecessary USB devices when recording or playing audio. Other resources for system optimization can be found at various web sites including TASCAM's.

Immediately after completing the driver and software installation, I made a test recording to see how much tweaking, if any, would be necessary, and was greeted with a badly glitching playback. Further investigation showed that some hiccups were actually in the recording, not just in playback. In fact, not even CD playback through the US-122 on either of my computers was glitch-free – not very encouraging. Since I use both of these computers with other audio interfaces, I know they're capable of better performance so I went looking for things to tweak that might be unique to this hardware. I discovered that reducing the size of the US-122's buffer (a driver adjustment) from the default setting was all that was necessary to make it play and record solidly. I expected that it might have wanted a larger buffer (and tried that first), but computer being computers, sometimes intuition doesn't work as expected.

While PCI and ISA bus audio hardware has been with us for years and their operation is fairly predictable, there are several different USB interface chips from which a computer manufacturer can choose, and some simply work better for audio than others. The manual (it's available on TASCAM's web site if you want to look it over before making a purchase) lists several USB controllers with which the US-122 has been tested. My desktop computer has one of the listed ones. The laptop's USB controller isn't on the list, but with nothing more than the buffer size adjustment, I got them both working smoothly.

TASCAM hints that some laptop users get more solid performance by using a CardBus USB interface instead of the computer's built-in USB por,t but to me this sounds like a desperate measure. While I'm confident that a good tweaker can get it to work on any computer that's not more than a couple of years old, I have

only a limited amount of patience with computers, and you might, too. Take this as a general caution rather than a potential problem with the US-122.

One dream I had for the US-122 was to use it as a mic preamp when recording with something other than a computer (I had my Creative Labs Nomad Jukebox 3 in mind). Since with the Direct Monitor switch engaged, there's a straight signal path from mic preamp to line output, it occurred to me that I could power the US-122 from a wall-wart USB power supply and use it as a portable mic preamp. Unfortunately this doesn't work. While the audio path is direct, it isn't completely hard wired. There's software-controlled switching that requires the computer's attention, and with only power connected to the USB port, no audio comes out. It's possible, once the US-122 shakes hands with the computer, to disconnect the data path (I hacked up a USB cable to try this) and the audio routing from input to output remains intact, but it really isn't a very practical way to extend the functionality of the unit. It would have been nice.

So How Are The Mic Preamps Anyway?

To answer Recording's most popular question, I set up the US-122, a Mackie 1402 VLZ-Pro mixer, my Soundcraft 600 console; and a Great River MP-2H preamp, recording my voice, acoustic guitar, and some small percussion instruments with an assortment of microphones. To eliminate the US-122's converters (and also to make playback comparisons easier), I recorded to my Mackie HDR24/96 recorder and listened to playbacks through the Soundcraft console. For the most direct signal path, the recorder was fed from the insert sends of the US-122, Mackie, and Soundcraft consoles. Microphones were AKG C-451 and Neumann KM84 small diaphragm condensers, Neumann U87, AKG C-414B-ULS, and Studio Projects C3 large diaphragm condensers, Beyer M88 dynamic, and Beyer M260 ribbon. The only variable was how consistently I could sing the same song half a dozen times.

Once a sample of each mic through each preamp was recorded, I juggled the tracks around so I could hear the same microphone through each of the preamps in sequence. To be brutally honest, (and given the price, you can just about expect this result) the US-122 was not first choice preamp with any of the mics. Equally significant, is that other than a shortage of gain when used with the M260 (which has rather low sensitivity), it was quite satisfactory.

Transformer-output condenser mics (the U87, KM84 and C-451) had some muddiness in the mid-bass area with the US-122 that wasn't present with the other preamps but this could be cleaned up with a touch of EQ. Transformerless condensers, (the C3 and C-414) didn't have that muddiness. but had the slightly brittle characteristic that's often associated with Mackie preamps. To be fair, both of those mics have an upper midrange boost which is typically exaggerated by a transformerless preamp. If you know what a Mackie sounds like with this type of microphone, you'll have a pretty good idea of the sound of the US-122. When

used with the dynamic mics (M260 and M88), the US-122 (as well as the Mackie) sounded noticeably thinner than when going into the Soundcraft and Great River.

In summary, the preamps are certainly usable within the scope of recording projects for which the US-122 is most suitable. It would be a whole lot less useful without decent preamps.

Beware the Phantom Menace

I discovered, by blowing out the input stage of the first US-122 I received for review, that it doesn't take too kindly to plugging and unplugging condenser mics with the phantom power switched on. It's always good practice to turn off phantom power before connecting or disconnecting mics, so, yeah, I should have known better, and so should you. The manual even says so.

So why did I mess up? I was using my laptop computer for bench testing since I don't have a computer in the shop. When connected to the laptop, flipping the phantom power switch on would cause the US-122 to lose its computer connection, requiring disconnecting and reconnecting the USB cable to bring it back to life. If the phantom power switch is on when the USB cable is connected, it starts up fine, including phantom power. Unplugging and re-plugging when changing mics got pretty tedious, so I threw caution to the wind, left the phantom power switch on, and ended up killing the US-122 when I hot-plugged a mic. Knowing that the full phantom voltage can momentarily appear across the input to the mic preamp stage if both pins of the XLR don't connect or disconnect at exactly the same time, many mic preamps have built-in protection against this sort of abuse. Apparently the US-122 does not.

The phantom power switching works fine without killing the US-122 when it's connected to my desktop computer, so I'm guessing that the power to the USB ports on my laptop is a little "soft," causing the US-122's power supply to drop below its working threshold while its 48 volt DC-DC converter comes up to speed. Incidentally, the phantom power supply adds significantly to the current that the US-122 draws from the computer. With phantom power off, drain from the USB port is about 180 mA, but it draws nearly 300 mA with the phantom power on.

TASCAM assures me that the US-122 had been tested with many laptops and they didn't encounter this power problem, but added that "laptops are different." I mention this here because its size, portability, and downright usefulness just screams LAPTOP PORTABLE RECORDING. If that's what you have in mind, try to test it with your own computer before you buy, or if you're going to buy a new laptop computer to use with it, ask TASCAM what they know works. Also, recognize that operating the US-122 with phantom power on will significantly cut into the operating time when running the laptop on batteries.

How's It Sound?

In a word, fine. The preamps are unexciting but certainly usable. The converters don't do anything nasty, and the analog input and output circuitry has sufficient headroom to handle full digital level with no sweat. There's no reason why you shouldn't be able to make perfectly satisfactory recordings with the US-122. Most of the recordings I made with it were live, direct to stereo using a Studio Projects LSD-2 stereo microphone. In this application, the preamp noise was completely overshadowed by the room noise, so it was no problem. If I was mixing 24 individually recorded unaccompanied vocal tracks, however, I'd wish for a quieter preamp. This is really my only sonic reservation, and I think that it would only be a problem on the most revealing recordings. It does what you'd expect it to do.

Incidentally, for some of those live recordings, I connected the Jukebox 3's input to the US-122's line output, recording simultaneously to the Jukebox and the laptop computer. It was neat to have the Jukebox to listen to on the way home, have a backup recording, and have the audio already on the computer and ready to burn some CDs the next day.

Wrap up

In summary, the US-122 is one of those "Why didn't anyone think of this before?" products. Much as I hate to use this trite phrase, it's an excellent value for the money. Why? Because it provides the functionality of several different pieces of equipment in a sensibly integrated package without any seriously weak points. You get just what you need for tabletop music production or live stereo recording: two mic inputs with phantom power, a mixer for monitoring without latency side effects, a headphone amplifier, and most important, a volume control for the speakers. Combine all of this hardware functionality with decent sonic performance, add in the versatile software bundle that's included and you have the heart of a compact and fully functional control room. Just add the talent and go make a record.