Congratulations on the purchase of the LANGEVIN STUDIO HEADPHONE SYSTEM. With this station you will be able to offer musicians a better sounding headphone amp than most major studios and be able to provide some significant improvements over basic stereo cue boxes or any other headphone system we know of. We subtitle this unit "THE MORE ME BOX" because this is possibly its most important feature. Along with the typical cue mix or the control room mix, the engineer can offer each musician a fader dedicated to their own instrument. Experience teaches us that each musician always wants to hear more of themselves and that trying to meet this demand with several musicians and with a few aux sends is quite a mind bending challenge. As long as each musician's headphone is plugged into a separate station, each can have their own custom mix within arms reach. This frees up the engineer to concentrate on recording and getting the best sound. It can free up console aux sends so that they may be used for effect sends. It also tends to allow the producer to concentrate on perfomances because the musicians monitoring needs are met quickly and easily.

While we strongly suggest that musicians not monitor too loud and possibly damage their hearing, we know that many musicians will tend to listen to incredibly loud headphones while playing. Most systems will distort or be gross sounding at these levels causing even more hearing damage, fatigue and headphone damage than this system will. It is designed to sound good at all levels from quiet to extremely loud with headphones of all impedances or efficiencies.

The communication features and options are incredible. This inexpensive headphone system has more and better communication features than any commercially available recording console or other headphone system. There are three main concepts.
1)   The ability that allows the producer to mute the music and talk or to talk within the music mix. The former is most appropriate during tracking with several loud musicians playing when the producer needs to stop a bad take. This is named <u>INT</u>erupt. The latter is more appropriate for vocal overdubs and coaching and is called <u>TALK.</u>
2)   The ability to talk to individual stations. This can prevent embarassment within a group, or allow a producer to only be heard by the conductor.
3)   The ability that allows the musician to talk and/or get the control rooms attention even if their mic is off during a playback or the room is dark. There is a built-in mic on the station with a button for "TALK". Pushing the button can light a LED (one per station) in the control room and *break into* the main monitors (if selected).

The Player can mute the headphone amps. The MUTE switch can be thought of as a "PANIC BUTTON" for those who have ever experienced sudden feedback or a weak signal suddenly "cutting in". These accidents generally mess with the mood and the hearing of the players. The Mute also functions to mute each side of the headphones. Primarily this feature is for musicians who wear only one side of the phones. This is common with vocalists, brass and string players and allows them to hear acoustically what they are producing. Functionally pushing the MUTE button cycles from ON (GREEN LED) to MUTE (RED LED) to LEFT MUTE to RIGHT MUTE (RED + GREEN = AMBER) and then back to ON.

Audio FADERS are used rather than rotary pots for the input levels. This gives obvious visual cues to the user. The MONO inputs PAN. The stereo inputs switch to MONO, SIM and STEREO. SIM is Stereo Image Modelling and creates a stereo image in the phones that is closer to speakers and reduce the difference the musicians hear when they return to the control room. This circuit slightly boosts the low and low-mid frequencies and makes them more mono. This is usually the best sounding setting with a stereo mix fed into the stereo inputs.

Tone controls. These are specially designed for headphones and musicians. The BASS control is designed to musically raise the loudness of the instruments BASS and KICK. The TREBLE is designed to raise the level of the HIGH-HAT and add a pleasant airy quality to VOCALS without harshness or hardness that often happens with headphones. Another design parameter was to help minimize differences between the most common models of headphones for studios. Technically, these are very gentle slope shelf EQs set to higher frequencies than commonly used. Center position is flat and very gradual near 12:00 on the control.

Individual external power supplies. These external supplies are not "wall warts". External power supplies create less problems, such as hum, in the stations. They reduce the weight and size of the stations making it easier to accomodate mic stand mounting. By having individual power supplies a studio does not have to rewire to have stations in iso-booths. It also increases system reliability because when a central power supply is a problem it is a central problem. Central power supply schemes also use the audio cabling and connectors which is not a recommended practice for valid electrical reasons.

Both 1/4" jacks and multi-connectors offer versatile wiring options. Balanced audio inputs and control logic combined with the recommended wiring schemes prevent the possibility of ground loops and hum. These inputs are fine with either balanced or unbalanced signals and +4 or -10 signals. Low noise circuits are used with plenty of gain range.

4 headphone jacks. While the unit was primarily designed with having a separate headphone mixer for each musician there are many situations where several musicians can share one station, such as harmony vocals. We could not fit a separate volume control for each headphone jack in the unit nor do we like the loss of damping factor and reliability that these attenuators cost. An external box with 100 ohm to 500 ohm 5 watt pots are cheap to buy or easy to build if needed.

Overload indicator (O/L) flashes 3 to 6 db below clipping and typically 10 db below audible distortion. However in practice when it is flashing the volume should be reduced.

The stations provide 3 audio outputs at all times. Two are used for a check of the headphone amps outputs so that the control room can hear exactly what the musician has mixed or messed up. The third line is the built in mic output and is not affected by the talk switch.

Simple, intuitive operation. No training or expanations to the talent is usually necessary. Just label the faders. Not all headphone systems are this easy to use. Very few musicians want to be trained to use your studio.

As LANGEVIN is the solid state branch of MANLEY LABS one can expect the unusual degree of high quality parts and construction. There are very few manufacturers of pro audio gear using this level of high quality parts. All Op Amps are Burr-Brown OPA2604 FET type running with 21 volt regulated supplies and were purpose designed for high quality, clean audio. The faders and right hand pots are ALPS and the pan pots are conductive plastic Bournes. Signals pass thru only one capacitor per phase and these are polypropeline. Resistors are all 1% metal film and most are 1/2 watt. The Power Amps are monolithics, are very new and designed to provide 150 watts peak and 60 watts continuous into 4 ohms per side. Because headphones usually require volts rather than watts, we limit the supply current appropriately. These amps are designed to never produce any click or thump or DC during power-up or power-down and thus will damage fewer headphones. They are fully protected against shorts, over heating and power supply faults. Best of all they sound very good (for solid state) and will drive headphones loud enough for drummers. These amps will drive speakers to reasonable volumes if needed, however watch out for over-heating (they will simply mute) and keep in mind that the stations were not designed for full power sine wave specs below 40 ohms.

Wiring requirements are only as much as needed. Different types of studios may only need or want certain functions. Multi-connectors are not needed for a basic system. The following sections give some of the wiring scenarios from the simplest to the most comprehensive. The simpler set-ups are typical of how many major studios are presently operating. Most of them are using one or two stereo power amps, lots of speaker cable and those boxes with a few volume controls. Each part of such a system will work but always sounds poor. They could work the way they are used to and improve the headphone sound for little cost if they gave more of a damn about the musicians performance needs than their lounging needs.

INSTALLATION

The most basic system only requires 2 lines from the control room to the studio. It is possible to use spare mic lines with the appropriate adaptors. Patch the consoles Cue Mix to the mic lines. In the studio patch from the mic lines to the stereo input channels. Use the console's TB (Talk Back) to talk to the talent.

Sometimes you have to set up a "Room Mic" or dedicated mic so that the talent can talk to the control room.

A better method is to use real line level cabling rather than mic lines because the mic signals are quite low in level and the above technique may tend to crosstalk the cue signals into those valuable mic signals. Do not use any speaker cables for line level signals because they are rarely shielded and will hum rather badly. Do not use the power amp outputs to drive the station. At the best it will sound lousy - at the worst you will blow up the inputs of the station.

If you have more lines available you can send more channels to the station. First choice would be a channel dedicated to the musicians own mic. The best point to take this from is the output of the track you are recording onto. That way they not only hear the mic but also the playback from tape. "MORE ME" is a common request and in itself is an improvement over typical cue systems.

The next variation is a simplification of the first two methods that is useful when the talent is likely to use several tracks or work on several songs. Use the main stereo mix for the headphone station's stereo input. Then use a mono aux send that is just the new tracks to be used for the "MORE ME" channel. The advantage is that the stereo mix is only one mix to set up, not two. In general the stereo mix that the engineer is hearing will be better balanced than the CUE mix. The artist usually is happy with this as long as they can have extra "ME" to zoom in on their performance.

The next variation is even "MORE ME". Add another line that represents the mic (after the mic pre). Maybe a stereo feed of their reverb would be a nice touch. Have you got more lines available? Some singers like the bass and kick on faders. The bass tends to help with pitch while the kick helps with phrasing. Now they almost gotta be happy. The only complaint should be either the air conditioning or the cigarrettes.

If a few musicians require phones and they can compromise a working volume simply plug up to four headphones into the station. If you expect that they need different volumes you can use a commonly available and cheap box with individual volume controls. Better yet is to use several stations. The engineer can send each station as many inputs as there are lines available. Again a better option is to use multi-cables to link stations. This saves on lines needed and simplifies the set-up. This method gives each player their own mix, volume, tone controls etc.

In larger set-ups some consoles have more busses than are usually required. One example is with an SSL console. With 32 busses and 24 track machines many engineers could use those last 8 busses to feed the stations. This is one easy way to pre-mix any number of tracks down to 8. The downside is that the Talk-Back does not send to these outputs unless the Slate button is pushed. A better variation is to use 6 or 7 pre-mixes and one channel for Talk-Back. You can access this signal from the Cue outputs. You may want to dead-patch ahead of this point so that the stereo cue send is available for effects. Then again, you could set up a pre-mix on the CUE mix and TB will be there for free and no tricky patching is required. These types of set-ups are typical with many consoles and a little experimentation goes a long way.

Most studios are pre-wired to accomplish any of the above set-ups and if not the changes hopefully are minor and easy. Anywhere from 2 to 24 extra lines that are run from the patchbay to the studio will do the trick. Some may elect to have one or more "25 way D" panels in the studio and only use multi-cables. This is a neat and organized style and a good way to wire a new studio. It may be difficult for an existing facility but not all that necessary because the 1/4" inputs are pretty easy to deal with.

The communication features built into the stations have not been addressed yet but if you can access 12 or more lines between the studio and control room much is possible. With most comprehensive systems some thought and work is involved in a complete installation in order to get everything the system is capable of. The following examples are for more advanced applications.

ADVANCED INTALLATIONS

To access the communication features the studio must use the 25 way "D" connectors because that is where the TALK-BACK signals are. The multi-cable connecting to each station should be 12 pair balanced lines with each pair shielded. 8 of these pairs are used for the 8 inputs. 1 is used for the TALK-BACK signal from the control room and 1 is used for a DC voltage that tells the station(s) to use the TALK-BACK signal. This DC signal can tell the stations to replace the music with the TB or to use the TB in the headphone mix.. The DC signal can also be used in advanced installations to allow the producer to talk to one station only. It also allows footswitch "TALK" and the possibility of a lit LED corresponding to the station when a musicial pushes "TALK".

The minimum wiring requirements to use the TB input is 1 shared audio line with the TB signal on it and 1 line shared for the DC signal. The TB signal is the producer or console mic signal amplified up to line level. If a push button switch is wired to simply short the two wires of the DC line all stations will hear the producer in the music. A more interesting way to do it is have two buttons - one for "TALK" (with musc) - one for "INTERUPT" (kills the music and replaces it with the producer mic only). Here one of the DC wires is switched to a negative voltage or switched to a positive voltage. This gives either "TALK" or "INTERUPT" to all stations.

The easiest way to hear the musicians is to set up a "Room" or "Ambiance" mic. Another way to hear the musicians when they press "TALK" is to monitor the shared TB signal. This line is shared between the producer mic and all the stations so be aware that unless the control room monitors are quiet you will likely get feedback when the producer talks. We do not recommend using this method unless the control room is monitoring with headphones. There are better methods and we will get to them soon.

Let us suggest the idea of <u>"D" connector Cue panels</u> with 1 "D" connector per station. With a little effort all of the features of these stations become available. We can have each of the stations mic outputs appear at the patchbay. These are unswitched so that they could be used so that the musician does not have to hit the button to be heard or for other tricks like ambience mics. The former is handy because most musicians hands are full and/or busy much of the time. The control room still is likely switching these mics but this is normal practice. With these individual "D" panels the control room can monitor exactly what musicians have mixed for themselves. All it requires is more patchbay points and more wire. Then we should have some logic circuits so that all this TB and monitoring is switched in when we need it and not just points at a patch bay.

At this point we should notice that the the wiring and panels are getting ambitious. Langevin will be introducing a few accessories to help fulfill the promise of the system. The first would be a that "D" panel in the studio with a few tricks to reduce the wiring demands. One idea is to select the station monitor lines and mic lines at the panel. Another idea is to mix those mic lines across a stereo pair so that it is possible to hear everybody at once. That reduces the number of audio lines running from room to room.

We could supply one or two panels with 4 "D" connectors and a few extra outputs for those Studio Loud Speakers (SLS). One panel would need a 12 pair cable and a logic wire thru the walls. Two panels would require a another logic line maybe. We would connect one panel to the other with a 12 pair cable plus that logic line. A bit easier.

Back at the console another Langevin accessory would be a small box with several communication buttons and a mic. The bigger buttons are labelled "TALK", "INTERUPT" and "LISTEN". Two rows of smaller buttons are labelled "1","2","3","4" and SLS. Next to the small buttons there would be two switches - TALK/INT and MIC/ MON. Pressing "TALK" allows the control room to talk with the music, INT kills the music but not the mic. "LISTEN" causes a relay to switch from the normal monitor source to the selected new source. This new source is selected by the MIC/ MON switch and the buttons 1-4 or SLS. If MIC then you can listen to any of the stations built in mics or all of em' in SLS. When a musician presses "TALK" at their station automatically LISTEN + MIC light and a LED corresponding to that station lights up. In other words you hear him or her. That other row of 1-4 + SLS is to talk to individual stations. That switch marked TALK/INT is to choose how you want to address them. This box is connected to the panels with a modest cable. Do you want 1 or 2 of these console boxes? 2 would give access to everything in an 8 station system however, in reality, it is unlikely that you really require "everything". A better reason for a commercial studio to have two boxes is that the producer and engineer

could have separate TB boxes. This is appreciated. There are also infrared remotes available that could be interfaced to one console box. Now the producer can pace and talk more. The "console box" can also be interfaced to the consoles TB system by a good tech. There are points available to access the basic functions. There is even points to be attached to the console oscillator. Turn on the oscillator and the phones turn off. Sort of a no smoking feature....

The patchbay can be wired any way you like but on the "D" panel solution you could have points for the 8 inputs, 1 TB, 2 SLS and perhaps 2 "station lines" that are selected mics and station monitors.

If one is wiring this system from a patchbay to the studio please consider running more lines than the 13 lines described. You may want to have each station mic show up at the patch bay. Add lines and points for this.

Another example, you would like to give 4 musicians a 'MORE ME" signal on channel 1 and still have the other channels for signals common to all stations. One answer might be to wire the patchbay with the 8 usual shared inputs and 4 or more points that are meant to be these "MORE ME" signals. You could wire the output of the shared channel 1 patch point to the "half normals" of these 4 "MORE ME" points. The shared sends are wired in the studio to all of the "D" connectors in the studio and the 4 individual lines are only wired to the "D" connector corresponding to that station. This requires a 16 pair cable instead of 13 and 4 more patch points. An 8 station system with 1 channel patchable individually is a 20 pair cable. 2 individual channels per station for 6 stations is 24 pairs (23 used) and for 8 stations is 27 pairs. These are common cable sizes providing uncommon versatility.

OTHER INSTALLATION CONCEPTS

If you have mostly understood the preceeding sections, you can see that a modest system is an improvement over your present system and that the full blown systems would deliver more features and performance than you have ever known headphones could have.

Lets consider that you really want 5 or 6 stations and you feel that two panels and 2 console boxes is a bit of overkill. You're right - you can always daisy-chain or LINK the stations. Each musician can still have a private mix and volume. When you talk to the first station in a chain the others will hear. Usually, you only need to talk privately with one person out there - don't chain that station. With Linked or chained stations you lose their stations mic but their TALK button will tell the system to LISTEN to the first station in the chain. No problem if they are situated near each other and that would be likely if they are Linked.

How about a major studio with sessions involving 30 musicians. Don't throw away the little boxes with a volume control for each musician. These Langevin stations will feed quite a number of headphones and sound better than most head phone systems with one or two big power amps. You will still probably only give them one or two stereo feeds. The SIM and MONO settings are useful here as is the tone controls. The MUTE feature where one side can be turned off is very useful with some string and brass musicians. You also get a separate TB to and from the conductor. These session players often bring in their own personal battered up phones and ear pieces. We gain some reliability with a distributed system rather than a single amp. With 30 players getting scale plus reliability is pretty important.

Speaking of ear phones, this is a major trend for stage monitoring. A clever stage monitor mixer could use these stations as easily as a studio engineer and gain the extra volume and features that are built in. It could help calm down some of cryptic those hand signals if they have a little mixer like this within arms reach.

The only things you have to be warned about are few. If you use both the 1/4" jacks and the "D" and you should be aware that one connector does not somehow stop the other. The 1/4" jacks are not "inserts". They are connected directly together which can be good for daisy-chaining but bad if you end up connecting two outputs together. Usually the result is both signals suddenly distort. For example one output is a console aux send that is coming in on channel 1 thru the "D" and you want to plug in the output of a guitar effects box into channel 1 thru the 1/4". Some studios will want to set up the patchbay so that points not patched are shorted to reduce noise (this is good). Once again that guitar effects box will distort as it is driving into a short. Here you just need to put in a patch cord into that one point to open the short or, in other words, dead patch it.

One reasonable thing to worry about with any Cue system is feedback. It can hurt an otherwise friendly musician with loud phones. It can even happen when a musician hits the TALK button when the volume is very loud and the phones are not well sealed. Feedback can be complicated due to having several communication mics in separate rooms being switched into control room monitors, Studio Loud Speakers and headphones. The remedy as always is use your ears to be aware of hints of feedback and be ready to turn the volumes down. One suggested practice is to grease pencil mark what you have found to be good settings on the TB mic and SLS. These controls are the hardest to be aware of in the control room because we rarely hear them there thus they are most likely to creep to Murphy's settings. If the headphones are simply on the verge of feedback into the vocal mic the only answers are to have the musician turn down their volume or wear better sealed phones or both. Occasionally you can insert a good (graphic) EQ into the "MORE ME" channel to notch out a frequency but usually the frequency changes as they change their distance from the mic.

Another typical CUE problem is leakage from the headphones into the mic. It can happen with quiet instruments and loud headphones. Leakage often is a problem with vocals and more so with harmony vocals. The usual fix is in the mix. We use gates and automation and phase reverse tricks and plenty of time to try to eliminate the leakage. The best solution is not to record much leakage in the first place. Try the same methods we suggest to prevent feedback. Less volume and better sealed phones. Personal "In Ear Monitors" should be encouraged in the studio. Some of them sound damn good. The only problem is that each musician should really invest in their own set because of health and hygene concerns. A studio can get some with disposable parts. The better personal ones are custom fitted to each ear. They help reduce the outrageous listening levels because they seal out much more of the external noise. This helps the musicians hear and hear longer. They also reduce the listening fatigue that sometimes makes those last hours so pointless. Just as they seal out external noise, they seal in the cue mix and drastically reduce leakage and feedback problems. A few hundred bucks are spent on In Ear Monitors and a few hundred bucks are saved in trying to clean up tracks on a console with gates on every channel. Looked at it that way, you get clean tracks for free and the musician gets to keep some good sounding phones and their hearing to appreciate them and your mix.

If you have to use click tracks, "In Ear Monitors" are the best solution. Much less leakage into the mics which is always a real challenge to clean up. Try a drum machine high-hat pattern with "swing" rather than a loud metronome-like click. With todays technology of MIDI tempo tapping and tempo maps there is little reason to even consider a click track. Some project studios still think that Clicks are the key to a tight feel. More often the click track is responsible for damage to what could have been a great human feel. Studios either kill the drummers natural feel or attempt to lay down real drum tracks after initial tracks rather than at the same time. Remember that music can be defined as people playing instruments together. If it wasn't for overdubs and iso-booths we could record great music without headphones and these stations would not be as necessary. Feel free to try it sometime - it's more fun and the results can be worth the lack of effort.

On the topic of hints - here's a few things to try sometime. Some singers and drummers have a real problem with headphones. If these artists are veteran stage performers they are probably more comfortable with stage monitors. You can set up speakers instead of headphones if you are careful and do it right. First you need a pretty dead room - across the spectrum not just the highs. Next set up the speaker at the dead

side of the mic - behind it if you are using a cardioid pattern or directly to the sides if you are using a figure 8 pattern. The figure 8 pattern is interesting because there is more of a proximity effect (more rich lows), less room sound, and you can use two speakers in stereo. You can drive the speakers with the Langevin station if you have a cable with a stereo phone jack on one end and bare wires on the other (don't short the tip and ring or sleeve). Set up a cue mix with only essential tracks that are very unlikely to be redone or not mixed. Try to mix in as little of the mic as possible. That keeps the overall volume reasonable. Keep in mind that some leakage will occur so have the singer be as close to the mic as possible and record thru the whole song keeping levels consistent. Lastly compression is to be avoided. Save the gates and compressors for the mix.

Some drummers just gotta feel that bottom to get the groove. If you set up a pair of normal speakers and send a cue mix to them nobody will be happy. Most studio type speakers won't get loud enough and the engineer has leakage and a corrupted room sound. Try a combination of headphones and speakers. The trick to using the speakers is just to use the kick and maybe snare and then roll off all the top and probably the most of the mids. Set the volume of the speakers by listening to the room and then the room mics. You will have some leakage but if you set it up right the leakage will sound good. We have known records made with a good sized sound reinforcement system reinforcing the drums in a live room. It sorta works.

Another drum trick where these stations are useful is in the mix. We set up a snare drum in a live room without the rest of the kit. We place a pair of drum sticks about 4 inches apart across the rim. Then we put an Auratone facing down on the sticks. We mic the bottom of the snare and the room. Then we feed the original recorded snare into the Auratone. We mix the mics we have set up with the original snare and smile. The Langevin station will drive the Auratone loud enough and you have control of level and tone while you are setting up the snare acoustically. Besides you are probably all set up to feed the stations easily and you don't need to drag in a power amp and a pile of adaptors. This trick works wonders on both tired sounding snares and drum machines. No one will stop you from EQing, gating and automating this once you have gone this far.

A similar mix trick is useful on some synths, guitars and vocals. Use your Cue system to send some sound back into the studio and mic the room. If the room is live you have the instant live chamber. You might even use the station's built in mic. If the studio is deader this trick may still yield some magic character because you still get speaker and room sound. You can even drive it till it distorts but watch out for the station overheating (it mutes) just as you lay the mix to DAT. Play around with delaying either the send or returns. Delays from a few milliseconds to 60 milliseconds are normal. If the room is not too live try adding some digital reverb to the sends. The room should add a touch of realistic stereo spread to the digital reverb. Experiment if you have the time.

The last hint is the most important. If you expect that you will be recording 5 musicians how many headphones and cue stations should you have ? Probably at least 8 pairs of phones and maybe 10 or 12. Headphones get abused in studios. Oprah could do a program on it. The phones get stepped on, the cords get yanked, and the transducers burn out. Most studios have a few broken sets in a box or a box full of broken phones that rarely seem to get attended to. Musicians usually avoid mentioning that they broke a set. All this adds up to the occasional shock when you thought you had enough pairs. Unless you check each set before the session, we guarantee the occasional nasty surprise. Do you have a spare headphone station for emergencies? Did you remember the headphone needs in the control room? Most producers and engineers like a method of hearing what the musicians are hearing. Some producers coaching vocalists wear headphones in the control room and sing into a real mic with the vocalist in the studio. Sometimes the vocalist is in the control room and everybody has to wear phones. All these methods work if you are ready for them. One place a session can avoid at least one pair of headphones is when bass player or guitar player play from the control room. Then you probably need real good set of big wall monitors and that's a topic for another day.

During the mix, don't forget to set up a station and a pair of good or trustworthy headphones (Grado builds phones for engineers). Because many record buyers only listen to phones this means you

gotta check your mix on phones if you care. You may be pleasantly surprised how useful they are when you need to set up a subtle effect and the effect device is behind you and out of the sweet spot. Good phones help zoom in on some tricky balances and layered effects. The Langevin station is great for these applications because it is easy to MUTE when your not using your phones, it has the MONO, SIM and STEREO switch on the stereo channels and you can check either side of a mix thru the mono inputs, and it has balanced inputs that are probably wired to the patchbay by now. Because it has multiple inputs you can "cue up" tapes or samples quite a bit easier and faster. Come to think of it, you might want a station in the machine room, the lounge, and the office and the ......

Thanks for reading this and thanks for purchasing the station or system. Don't monitor too loud if you can help it.