

# Student Assistant Techs

Student technicians are responsible for four main areas of expertise in our concert series, for external clients and providing recording services to students. They are:

- Audio Recording
  - Concert archive recordings
  - Audition recordings
- Video Recording
  - Concert archive recordings
  - Audition recordings
  - Special projects (promotional/recruitment)
- Projection
  - Stage Lighting design
  - Multimedia projection
  - Screen set up
- Live Sound Reinforcement
  - Audio playback
  - Voice and instrument amplification

Some things can be learnt in a single session, and some things need to be structured in a progressive way. Projection is easiest area to master, with audio being second, live sound reinforcement third, and video being the hardest (not least because it requires having mastered audio recording too). Live sound isn't really hard, but it is high pressure, often requiring quick thinking, so we'll consider it to be harder than audio recording.

Your training will take you through each area of expertise as your time commitment allows – the more areas of expertise you have, the more shifts you'll be able to take.

Support is provided in all areas throughout your time here – you are always welcome to ask for advice on how best to tackle a particular task or job.

As you progress through the training for each area of expertise, you'll have opportunities to use your skills in real performances. We audio record ALL Music at Memorial concerts, video record at least four per semester and handle multiple audition recordings for students – these normally consist of soloist and accompanist, and generally use a more complex set up for audio than the typical concert set up but a simpler video set up. Live sound gigs are less common with around 10-15 per year, ranging from a single microphone, to miking up an 18 piece band. Projection set up is even more rare, but thankfully, very easy to learn. Special lighting set ups are also quite rare, but straight forward. You'll build your confidence, abilities and experience as you progress and while you might not always be inside your comfort zone, we'll never push you further than your capabilities and training have taken you.

What we need in our techs is attention to detail, sound listening and seeing skills and a good attitude.

## BEFORE YOU START – CORE RESPONSIBILITIES AND OBJECTIVES

There are several reasons to have a list of rules, responsibilities and objectives. They safeguard the school's reputation and investment in equipment, and help to avoid awkward situations for you among your peers and the school's clients.

At the basic level, being a tech is about safety for you, the performers and the audience. At the advanced level it is about having a concept of sound reproduction – this is a subjective thing that can only be learned through experience, exposure and exploration. Regardless of what level you've reached in the subjective world of good sound production, you must think about safety and security of equipment, and everyone who comes in close proximity to it.

1. Treat all equipment with care and respect – equipment is expensive to replace, and in many cases, the school is unable to sustain a replacement program of equipment that gets damaged beyond repair. Equipment is intended to work effortlessly as part of the overall system. If you have to force something, chances are good you're doing something wrong, and chances are good you're going to damage something. So STOP, and think about what you're doing. Microphones are particularly delicate. Treat them with care, and whenever you're handling them, put your concentration into what you're doing. Not concentrating can easily result in drops which are usually fatal to microphones. Most of our recording microphones are worth over \$500 each, some are worth more than \$1,000.
2. All sessions must be booked through the Concert and Facilities Coordinator. Even pro-bono sessions must pass through our booking system so that we can keep track of room and equipment usage. I would recommend not getting into the habit of offering free recording services for friends because it can easily put you in an awkward position of having lots of work for which you're not being paid. However, it is acceptable in isolated cases. You must not in any circumstances get paid directly by individuals for service unless you are using ALL your own equipment to do so. If you're using the School's equipment and facilities, you must be paid through the student assistant program.
3. If equipment gets damaged, report it immediately to the Music Technologist. If you discover damaged equipment, report it also. We can't fix it if we don't know it's broken! All damaged equipment creates a hazard.
4. Be especially careful of trip hazards when setting up. Cable runs must be tidy and secure in order to avoid trip hazards. Avoid stretching cables and running them where they are not able to completely trail on the floor.
5. Don't phone it in... if you're not trying to create the best quality product you're capable of, you're doing a disservice to the people you're working with and to the School's reputation.
6. Know what service the school of music has agreed to provide. If you are being asked to provide something that is out of the scope of the services we offer, or that jeopardize safety by not utilizing enough people to do the job properly, address this with the Music Technologist or the Concert and Facilities Coordinator.

# Four track audio recording in D F Cook Recital Hall

## EQUIPMENT YOU'LL BE USING

**Steinberg MR816x** – the MR816x is an *audio interface* - an essential part of any computer based audio recording set up, and is the component that allows us to send our audio recordings from microphones to the computer, and from the computer to our ears. The MR816x is like a high quality sound card.

**Neumann KM184 microphones** – hanging from the ceiling in Cook Hall, just in front of the stage, are two Neumann microphones set up in an arrangement called ORTF. ORTF is a very commonly used stereo miking technique for ensemble recording that gives us a realistic representation of our sound stage.

**Neumann KM183 microphones** – hanging from the catwalk outside the soundbooth are two Neumann microphones set up in a spaced arrangement, commonly referred to as A-B. These microphones are placed to pick up the audience applause, and the ambient reverberation from the hall.

**Cubase** – Cubase is a DAW (Digital Audio Workstation). All computer based recording systems are called DAW's. We use this to record our audio, and to edit and mix in Cook Hall. In Suncor Energy Hall, we use Nuendo which is Cubase's brother. They look almost identical, and work in the same way. Their files are even interchangeable, so a project started in Cook Hall can be finished in Suncor Hall and vice versa. These products are so alike, they are affectionately known throughout the audio industry as Cubendo (Cubase-Nuendo...).

## OTHER IMPORTANT TERMINOLOGY

Before we move on, let's just go through a quick description of terminology you may come across in the text.

**Stereo Pair** - refers to the two microphones hanging from the ceiling in an ORTF formation.

**Monitors** - The speakers that we use to listen to material being recorded or played back through the recording system. Any reference to monitors in this article refers to the speakers. The computer monitor will be referred to as the screen.

**Bounce** - this is the process of mixing down several audio tracks in Cubase to a stereo (two track) file, which makes it compatible with CD players, MP3 players and any other media player.

**Patchbay** - the central point at which we connect our recording equipment to our microphones. There are patch bays below the stage, in the sound booth and at the back of Cook Hall.

**XLR** - A standard connection for audio components in a recording system. It is used to connect



microphones to our audio interface, and our audio interface to our speakers. XLR cables are also used in live sound set ups. The connection has a male and a female end and they click together. The male end has pins, while the female end has holes. Normally, audio signals are SENT through a male connection and RECEIVED through a female connection, though this isn't always the case.

**Clipping** - When the signal reaches a point called zero, it has clipped. Digital audio is a description of audio characteristics as a number and unlike analogue, cannot physically exceed zero. The result is that anything that goes above zero gets cut off, and this is heard by significant distortion to the sound which can't be removed in editing. So, it needs to be avoided. Use the metering within Cubase to determine correct levels to avoid clipping.

## HOOKING IT ALL UP

Connecting the components together is straight forward, and most of the time, will already be done for you. The only component which might not be ready to roll is the connection from the microphones to the Steinberg MR816x. If the microphones have been used with another system prior to your arrival, it's possible that they are no longer connected to the MR816x.

The picture below shows where the connections come into the sound booth from the hanging mics. They're on the left wall as you come into the sound booth. There are only four connections on the really old blue metallic rack have any relevance to us. Those are labelled STEREO PAIR (labeled in this picture as ORTF Mics and CATWALK MICS).



When I explained what XLR connections are above, I said that MALE normally sends, and FEMALE normally receives an audio signal. Here is one of those exceptions! We are expecting the wall connector to be a Male connection, since it will be pushing the signal on, but it's actually a female connection. The sound booth was wired long before my arrival at the School of Music, and I have no idea why it was done this way. In order to enable us to connect to other equipment, we need to correct the gender of the connections using a TURNAROUND cable, or a GENDER CHANGER. This gender change, can be made at any point along the chain of cables, but is at some point necessary if we are connecting to any manufactured equipment.

The other end of the cable run here needs to connect to the back of the MR816x - MIC/LINE INPUT 3 (left) and 4 (right) for the Stereo Pair, and 5 (left) and 6 (right) for the Catwalk pair.



...and this is the cable you use to connect. Note that it already has the male XLR connectors so you don't need to worry about a turnaround cable.

With the connections made, we now need to turn on the equipment, if it's not already on. It's important that you follow the correct order – partly to satisfy the finicky nature of it, and partly to protect the equipment and your hearing. Turning on the speakers before everything else can often result in loud bangs when other equipment is subsequently powered up. So, its most important that speakers get turned on last, and get turned off first. Failure to do so may permanently damage the speakers, and if you're too close to them when it happens, can also affect your ears. You've been warned! The MR816x likes to be turned on before the computer is booted up. If its already powered up when you arrive as is often the case, cycle the power (turn the box off, then back on again), to reset the device. The power button is located on the front right.

Next, turn on the computer. Once it has booted, you can turn on the speakers (power buttons on the back of each speaker). You may prefer to use headphones, in which case, plug them into the front of the MR816x just to the left of where you plug the headphones in. When you're finished, power down the speakers. Please leave the computer and interface turned on.

## RECORDING WITH THE MR816x

Above, you'll recall that we connected our stereo pair microphones to MIC3 and MIC4, and catwalk pair to MIC5 and MIC6 on the back of the MR816x. On the front of the interface, is a knob that controls the amount of amplification of the microphone signal – the pre-amp gain for each input channel. Experience will guide you to where these knobs need to be for various types of groups that you will record. The quieter the group, the more pre-amp gain you'll need. We aim not to change the level of gain during a performance, so you need to know what kinds of groups are performing, and adjust for the loudest one.

The good quality of our equipment means we don't have to push the pre-amps hard. If we record at a level that is too low, it doesn't matter too much... well, as long as it's not too, too low. If the meters in Cubase show clipping when you record, you need to turn these knobs down (you'll find out the signs of clipping as you read on...). The knobs for MIC3 and MIC4 should be at the same levels as each other when you're recording using the stereo pair to get a balanced sound that accurately represents what is happening on stage. Similarly, the knobs for MIC5 and MIC6 should also be at the same level as each other.

If you're wondering why we don't use MIC1 and 2 for our inputs, its simply because the connections for these two channels are on the front of the interface, and the cables are more likely to get in our way if we use these connections.

## CUBASE BASICS

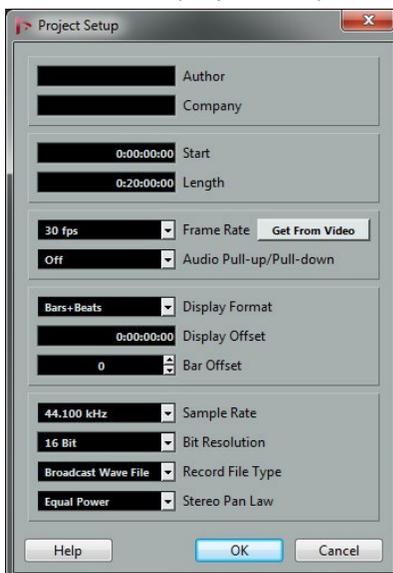
Cubase and Nuendo are very complex applications, and it certainly wouldn't be practical to go through everything. For the purpose of this session, we'll just go through the basics of how to set up a session and how to record a session. Later, we'll cover them in a little more detail, but this is all we need for the purpose of recording concerts. We'll just refer to Cubase from here on, but the same applies to Nuendo when you get to use that.

## SETTING UP

So, in the previous page, we went through the process of setting up the hardware. Now we need to go through the process of setting up the software.

1. Double Click the Cubase icon on the desktop, or single click the icon on the taskbar.
2. You will be greeted with an Open Document Options dialog. Choose New Project and click OK.
3. You will next be greeted with a list of templates that you can use as a basis for your project. Choose **Cook Hall Basic Concert** and click OK.
4. Next you need to choose a Project Folder. This is where all your audio files and project files will live. Use the D drive (called AUDIOFILES), and then your personal folder, then click the New Folder button.
5. Type a name for your project and use the following naming convention –  
YYMMDD – Project Name  
YY = Year, MM = Month, DD = Day  
For example – **120730 – Project X**
6. Now your project will open. You'll probably see a couple of windows including the main project window with a timeline and possibly a mixer window. At this point, your project is not yet saved. Go to File > Save As, then type a name for your project. It will already be in the folder you created in step 5, but for the purpose of easy finding again when you need it, give your project the same name (eg, 120730 – Project X), and hit Save.

Next, you need to check the project settings and make sure they are suitable for what you are doing. For most of what we do, you'll be recording using a Sample Rate of 44.1Khz and a Bit Depth of 16 Bit and the Basic Concert project template includes these settings. This allows us to put our files straight onto an



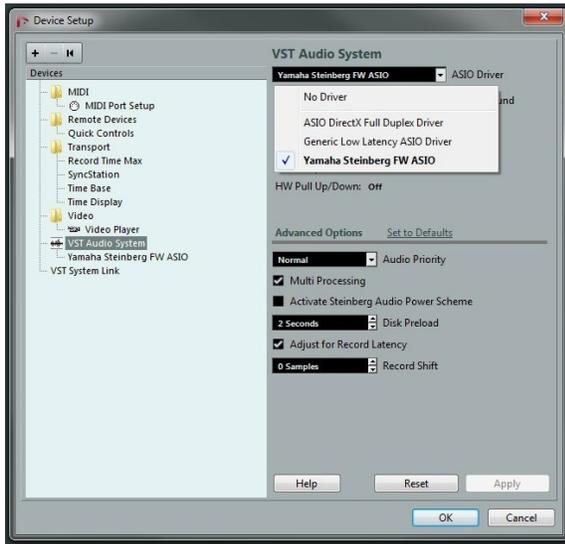
audio CD without having to do any conversion. Sample rate is the number of audio snapshots taken each second (in this case 44,100) and the bit depth is the resolution of dynamic range, or the number of volume steps available from quietest to loudest (in this case 65,536).

You can check/change this information by going to Project > Project Set up. Get into the habit of not changing these figures once you have started recording a session. This could give you some complicated problems to figure out later! If you record using the wrong settings, there are ways of changing it in post production that are better than changing the settings after you have started recording. We'll deal with that separately to avoid overloading you with information here, but just understand that these two figures are important for determining audio quality and that once you've started recording material, you should continue to record that whole session

with the same settings. We'll never want to use settings below 44,100kHz and 16 bit and doing so will severely compromise your audio quality.

Next, check the Audio Interface settings. Typically these don't change, and as long as you turned everything on in the right order, the interface settings should already be set. But there are several ways we can configure Cubase so it would be good to get into the habit of checking these every time you start a project.

First of all, we'll check that we are using the correct audio interface drivers.



You can do this by going to Devices > Device Set up. Under VST Audio System (selectable in the left side of the window), you should see the ASIO Driver – this needs to be set to Yamaha Steinberg FW ASIO.

Under the Yamaha Steinberg FW ASIO tab selectable in the left side, make sure Direct Monitoring is checked.

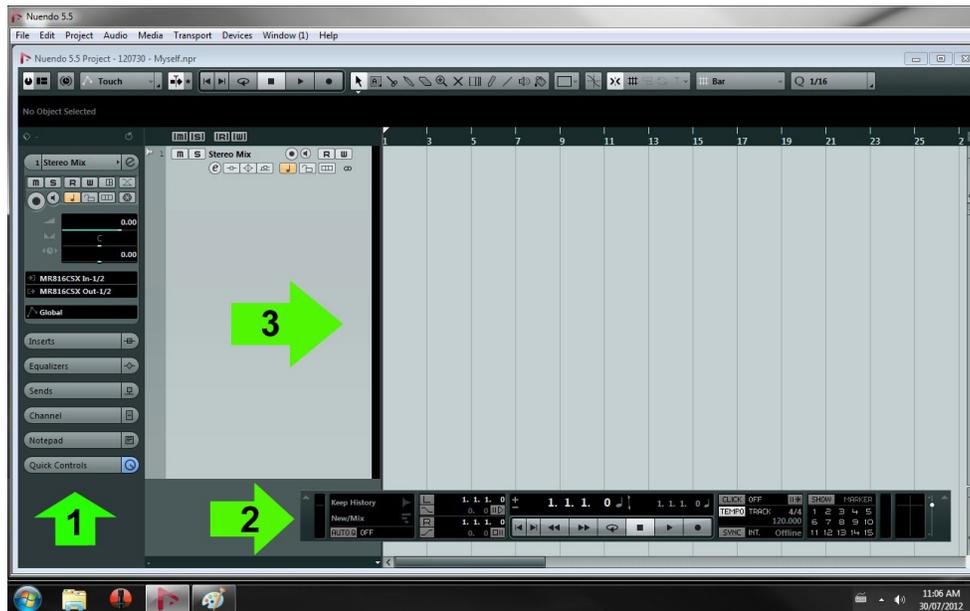


## WE'RE FINALLY READY TO GET STARTED...

There are two main windows that you'll use in Cubase. The Project Window and the Mix Window. The Project Window is where you see and edit the wave forms of the audio that you record. You can also use the project window to set up your recording channels and lots of other things.

The mix window is where you mix, or blend the audio tracks you've recorded. It is set out like a conventional audio mixer. It's more useful for multi-track recordings.

Lets take a closer look at the Project Window...



This is from Nuendo not Cubase, but the look is similar, and functionally they are identical at this point.

1. The Inspector. If you have a single track selected, you'll see a virtual channel strip for it called the inspector. It shows the inputs, outputs and gives you easy access to effects processing through Inserts, Equalizers and Sends. The Channel button shows you the mixer channel exactly as you would see it in the mixer window.
2. The Transport Bar. This is where you start and stop recording, rewind, fast forward and do several other tasks. Basic controls from the transport controls (Record, Play etc) also appear at the top of the screen.
3. An audio track. This is what you record onto. You'll see the audio waveform appear as you record.

For your set up, you will be using two stereo audio tracks to record the concert. To set that up, focus in on the inspector. Here's what everything does -



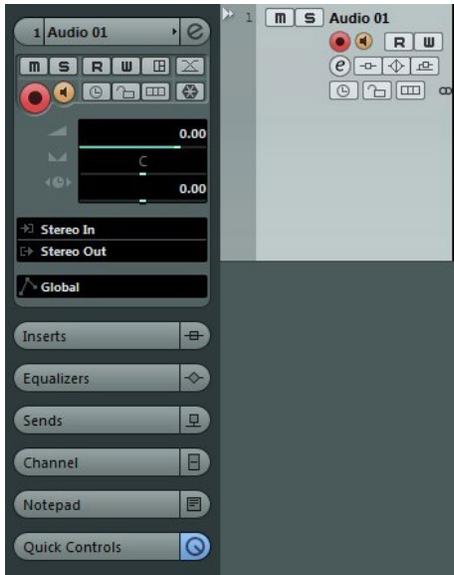
**Fader** – this will change the volume of this channel. It won't change the level of audio that gets recorded – that can only be done with the gain knob on the MR816x. This will change the level of volume that you hear through the speakers for this channel. It will also impact the level that gets recorded when you bounce down. We'll deal with that later.

**Panner** – this will change the position Left to Right. If you have two separate mono channels you'll have one panned hard left, and one panned hard right. With a stereo channel, there's no need to set this as it will automatically be panned to hard left and right. You can change this though if you need to, and you'll find out about that when you do more advance recordings.

**Input** – this determines where the sound for this channel comes from. If you set up as above, with microphones plugged in to inputs 3+4 on the back of the MR816, this will need to be set to MR816x –In 3/4 for the stereo pair, and 5/6 for the catwalk pair.

**Output** – this determines where the sound for this channel gets routed to. MR816x Out 1/2 is the main output. Output routing can get complicated pretty quickly, so for now, just know that you'll always want the output to be MR816 Out 1/2.

## GETTING READY TO RECORD



Once you've set up your channels, you need to record enable them so that when you hit record, audio will be recorded to them. Before doing this, it's worth saving again. Ctrl+S or File > Save will save your file.

The Record Enable icon is a circle, with a black dot in the middle, like in the image to the left. It appears both on the track itself, and on the inspector. It will be a grey/blue colour when it's off, and red when it's on.

Click this icon and it will turn red, and the track will be record enabled. Now, whenever you hit record, this channel will be included in those that record audio.

To the right of the record enable button is a monitoring button. Again, this is blue/grey when it is turned off. It will turn to a kind of orange colour when turned on. This should be turned on while you record, and off while you mix or listen back. Basically, it allows you to monitor the input of this channel while you're recording. If it's not selected, you won't hear anything you record on this channel. While playing back, you actually don't want to monitor the input. You want to monitor the output. So, turn the monitoring button off when listening back to tracks.

If you make a noise in the Recital Hall, the level indicators should show you that your two mic channels are working correctly. **If you don't get any sound, go back through the steps and make sure you haven't forgotten something.** Although Cubase set up is fairly automatic from session to session, it is possible that inputs or outputs don't set themselves up the way you need them. If you don't pick up audio, it won't get recorded. There is no way to get this back at a later date. So make sure it's working.

If you get a chance to do a sound check with the performers, you can trim the gain knobs of the MR816x to suit the performance. What you need to be able to do this is for the artist, or artists to perform at the loudest they will perform during the show, while you increase the gain levels until the music clips. Reduce that level by about 15% to allow for the fact that people often perform at a different volume than they sound check. I wouldn't advise going past the 1 o'clock position for a concert performance, because the applause itself will clip and while that's not as much of a big deal (the sound of applause is actually fairly similar to the sound of digital distortion), we should avoid it if possible.

If you don't get to do a sound check, here's a general rule of thumb for levels...

- Solo performers - 1 o'clock
- Small ensembles, except brass - 12 o'clock
- Accompanied singers - 12 o'clock
- Small brass ensembles - 11 o'clock
- Large groups (wind ensemble, chamber orchestra etc) - 10 o'clock

Always monitor your levels to see if you're in the right ballpark - remember that a quiet work will be much quieter than a louder one... it might sound obvious, but if the performers start with something quiet and you bring their levels up, you might find it clips when they do a louder piece later on. Since you're attempting to capture a realistic performance, you want to avoid changing the gain levels as much as possible once the performance has actually started. A quiet performance will see a much lower level on the meters than a loud performance.

## HIT RECORD

So, with your setup done, you just need to wait for close to the start of the performance, and hit **Record** on the main transport bar to actually start recording. If it's been more than a few minutes since you set up your session, give the transport bar a little extra time to start recording (if the hard drives have gone to sleep they require this extra time). It's always better to record too much than miss the start! You can edit out what you don't need, but you can't add in what you didn't record.

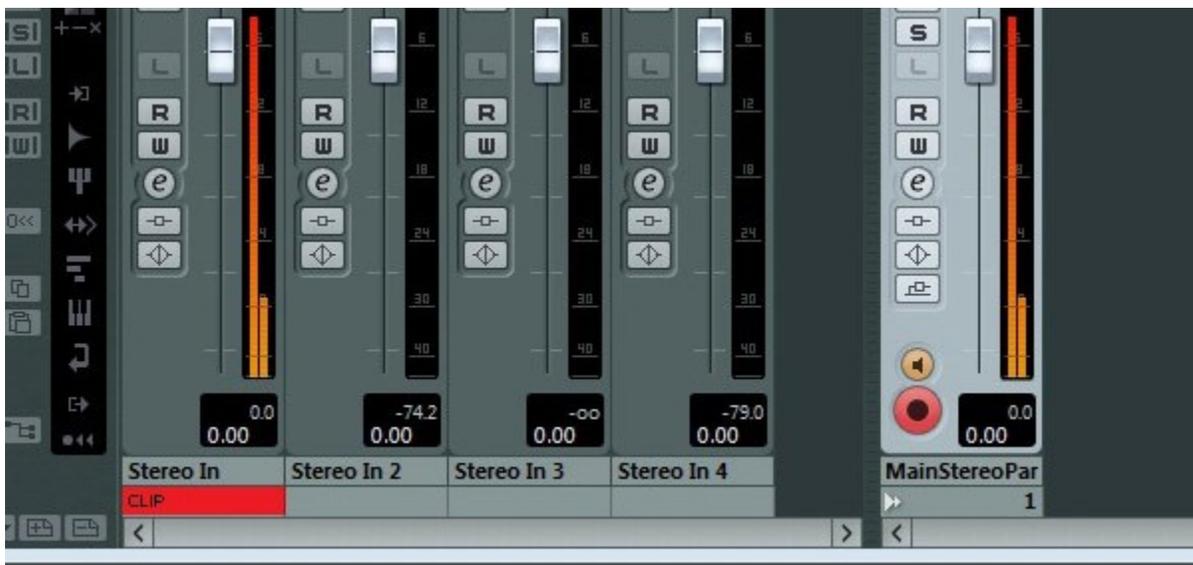
## WHAT TO WATCH FOR AS YOU'RE RECORDING

As you record, you should be able to see the wave form being created in the project window.



If you need to change the input gain levels, it would be worth putting in a marker so that you know where that happened. You can compensate the entire performance later. To put in a marker, press Shift-M. You can give your marker a name, or leave it blank, and then hit the enter button again.

You can use the Mix window to monitor the level of your tracks, and see when tracks clip. You'll see an indicator appear under any channel that clips.



You can reset the clip indicator by clicking on it.

A red clip indicator is okay if it's a fast transient like applause or a loud drum hit, but should be avoided where possible. If it happens more than twice, you'll need to reduce the pre-amp gain. If it keeps doing it, you definitely need to reduce the pre-amp gain!

## **AT THE END**

Hit STOP on the transport bar, and then press Ctrl + S or go File > Save.

To exit out of Cubase, hit the X in the top right corner. You may need confirm that you wish to save changes before it actually exits completely.

## **EDITING**

When we edit our regular concerts, we typically do a static mix. That is, without changes in our fader position as the concert progresses. This retains the original dynamics of the performance. To ensure our output file is just as it is inside Cubase, the master fader should be at unity (zero). You do all this inside the mixer window. You can choose the mixer window by hitting F4 or by going Devices > Mixer.

Back to the project window, and we'll first deal with tidying up the tracks. A typical workflow will be –

- Normalize your project
- Edit the length of your tracks
- Add fade in and fade out to each track
- Export your tracks

For very dynamic performances, you may also wish to use a tiny bit of compression/peak limiting. This requires some careful use, so if you're not sure about how compressors work, do not use this.

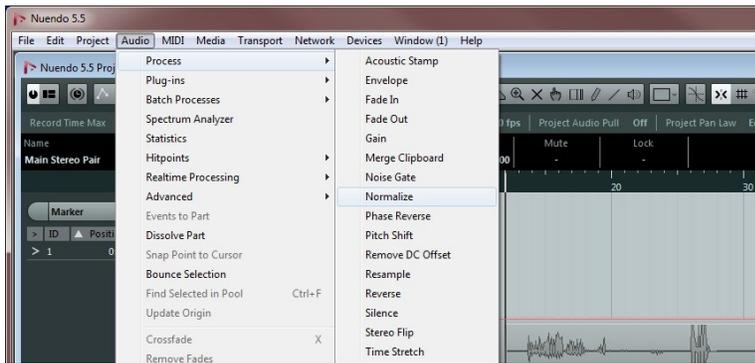
As an added first step, you may wish to copy your recording to a different track or use a new project file. Cubase works non-destructively with your files, so even if you mess something up, your project wave files can be restored to their original, however, the easiest way to get back is simply to edit from a copy. You can copy your audio by setting up a new stereo audio track (right click in a blank space in the project window and choose New Audio Track. Hold down the Alt key and drag your audio file from its original location to the new audio track. If you have multiple regions in your track, you'll need drag them all over in the same way. We'll edit from this one, so press the M button (mute) on the original track so that it doesn't interfere.

## **Normalize**

If you have multiple regions in your project, select them all and bounce them to a single region. You can do this easily by going to the very end of your project, holding down the mouse past the end of the project and dragging to the beginning of the project. Remember, do this just on the track you're working on. Now go to Audio > Process > Bounce Selection. We do this so that when we normalize, it normalizes the whole concert as a single event, rather than individually as dictated by the separate regions.

When prompted if you wish to replace the existing selection, say yes. This does not destroy your original file.

With your single file selected, now go to Audio > Process > Normalize...



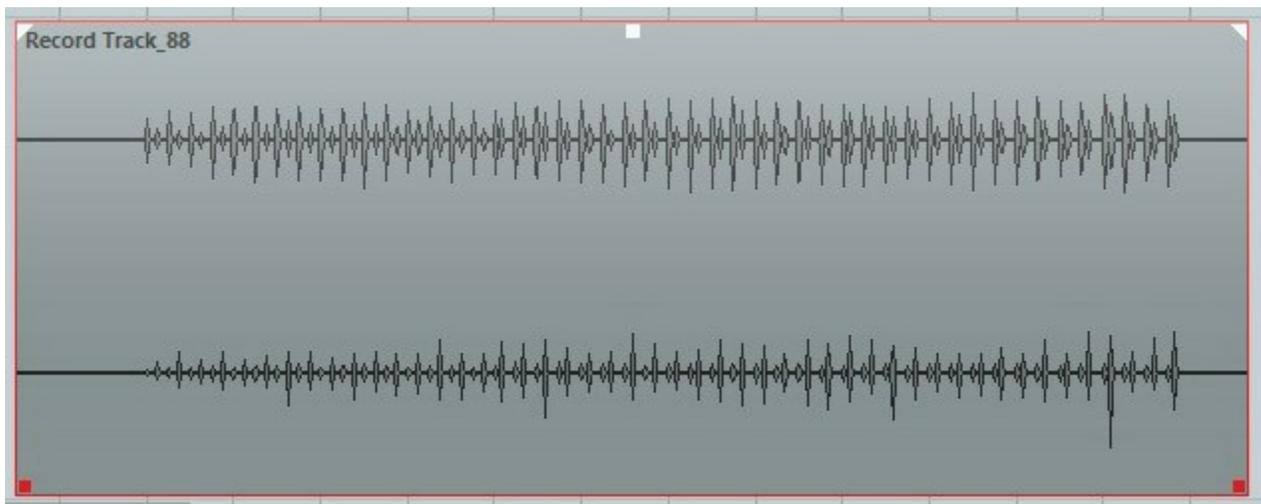
In the dialog box that appears, click on the black box, and change the “Maximum” to -0.5dB. Then click on process.



## Edit Your Tracks



Next you need to cut your tracks up to remove silence and audio that you don’t want on your final CD. You’ll cut your audio into regions which will represent the individual tracks of your CD. Use the scissor tool to cut your tracks roughly into regions by track/song/movement etc.



When you have a region selected, you’ll see five handles. Bottom left and right are squares – these allow you to change the start and end points of your region for accurate trimming of your audio tracks. Top left and right are triangles. Drag these to create standard fade ins and fade outs. The square handle top middle allows you to alter the volume of the selected track.

Edit each region so that the silence at the start is between 0.25 and 0.5 seconds. If there’s a breath or sniff at the start that’s important to the track, go 0.25 before that. Edit the end of

each region to allow up to four seconds of silence after the music has stopped. You may need to cut shorter than this so as to remove undesirable noises, but be sure to retain all the room decay sound. Make sure the edit sounds natural. You can add silence at the end in other ways if noise means you need to cut off before the four second point.

When you put tracks onto a CD they must start and end with a zero volume (infinity). If they don't, they will be heard as clicks at the start or end of the track. The easiest way to make sure this happens is to use fade ins at the start of each track, and fade outs at the end. Fade ins and outs need to be at least 10ms to be useful. That's not hard to achieve! You can create long or short fades depending on the needs of the track.

## Applause

There is no standard for dealing with applause out there in the real world. While some artists may like to hear their entire applause at the end of tracks, it can be tiresome to listen to if you're anyone but the artist! As a rule of thumb, let's work to the following standard –

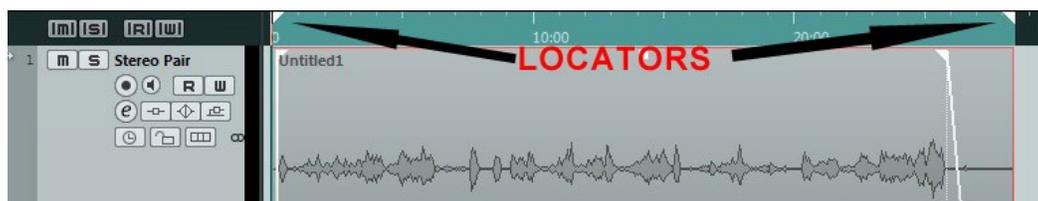
- Small audience: 5 seconds applause, 5 seconds fade out... 10 seconds applause in total.
  - o Last track – 8 seconds applause, 8 seconds fade out.... 16 seconds applause in total.
- Large audience: 10 seconds applause, 10 seconds fade out... 20 seconds applause in total
  - o Last track – 15 seconds applause, 15 seconds fade out... 30 seconds applause in total.
- If a performer asks you to keep all the applause on a recording, do so.

## Movements

Dealing with movements is tricky. Sometimes there are long pauses between movements, other times they practically go straight into each other. Use the performance as your guide, but make gaps even between the movements of a work, and never include more than 5 seconds of "silence" before a fade out. A gradual fade out is preferable to an abrupt one – you will hear the difference in the ambient noise. A fade of one second is good.

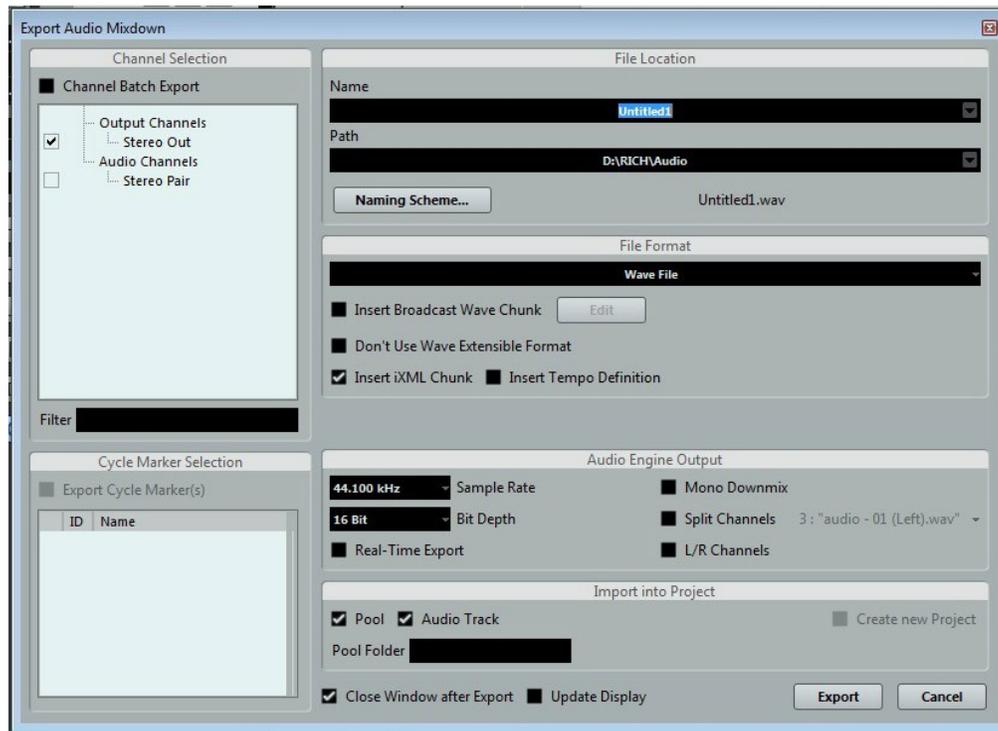
## Export

When you use the export function in Cubase, the region that you wish to export is defined by locators at the start and end of your selection. When selecting regions, you can easily place these locators by selecting the region then hitting 'P'. You will see the region selected appear in the ruler at the top of the project window.



When you have done this, you can manually move the start and end if you need to, by dragging the triangle locators, to provide additional silence at the start and end of your track. Just be sure not to infringe on the next region or this will become part of your export.

When you're ready to export, click File > Export > Audio Mixdown. A dialog box like the one below will appear.



Give your track a name (or number) that will be meaningful to you when you add the tracks to a CD listing (I typically use 01, 02, 03, etc...), choose a path (start with the Project Folder, and create a folder within that called CD tracks), and choose the file format (Wave File).

In Audio Engine Output choose the destination sample rate and bit depth – remember the sample rate will be 44.1Khz for and 16bit for CD. Uncheck the Pool and Audio Track boxes under Import into Project.

Click Export and your file will be created.

Repeat this export process for each track.

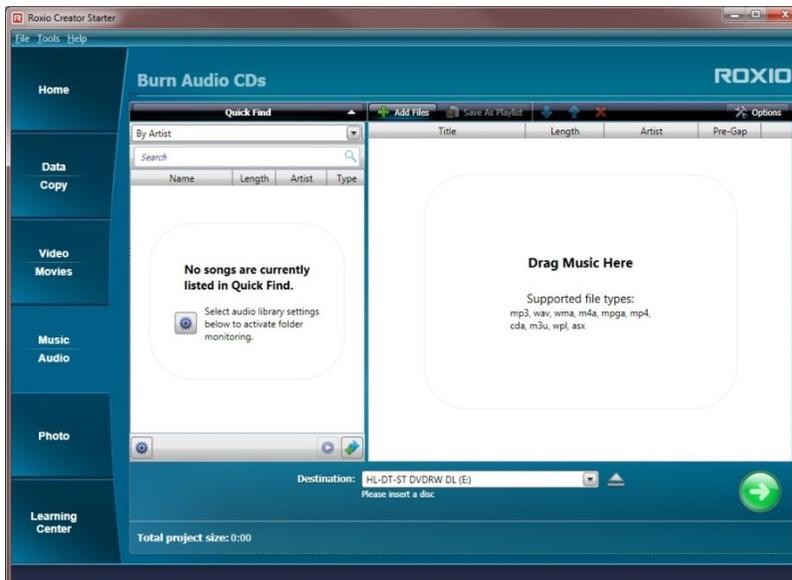
## CREATE A CD

Use Roxio Creator Starter to create an audio CD of your tracks.

Double click the icon on the desktop.



Click the Burn Audio CD's icon.



Click the Add files button and choose the files to add, or simply drag the files from where you stored them.

When you've added the tracks, you need to make sure the Pre-Gap is 2 seconds. This creates a better flow for the CD. Go through each track and make the Pre-Gap 2 seconds.

Check the order of your tracks, insert a blank CD and click the Green arrow to start the burn.

When you've burned your CD, listen to at least the start and end of each track to ensure everything is as it should be.

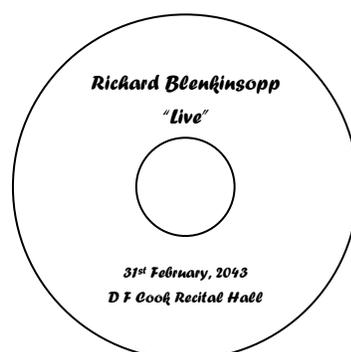
## CD RECORDING STANDARDS

School of Music recordings should have some kind of consistency, so I would like all recordings to be submitted with the following features -

- Each track should have a lead in time of no more than 0.5 seconds. When I refer to lead in time, I am referring to the actual audio file. A CD will by default put a 2 second silence before the start of each track, and while it's possible to override this, I would like it to be kept, and for the audio track to reflect that two seconds. A lead in of 10ms is enough to avoid playback problems, but between 0.25 seconds and 0.5 seconds is ideal. The start of the track may be considered to be the breath of a wind player, the sniff of a string player. Keep in the human element, but avoid oodles of silence at the start of each track.
- Normalize the entire performance to -0.5dB. Do not normalize individual tracks. If you have spoken introductions, these are likely to be very quiet. You can normalize the introductions separately - normalizing to -12dB should give you a good bit of boost without introducing too much background noise.
- Applause - Small audience: 5 seconds applause, 5 seconds fade out... 10 seconds applause in total.
  - Last track – 8 seconds applause, 8 seconds fade out... 16 seconds applause in total.
- Applause - Large audience: 10 seconds applause, 10 seconds fade out... 20 seconds applause in total
  - Last track – 15 seconds applause, 15 seconds fade out... 30 seconds applause in total.
- Applause – special: If a performer asks you to keep all the applause on a recording, do so.
- Include two seconds of silence at the end of each track.
- Split works that are in more than one movement except where doing so would cut off the audio. This would be the case if the performers start the movement before the decaying room sound has finished for the preceding movement. Don't include long silences or tuning up.
- Complete your editing of a performance within one week of that performance, and hand it to the Music Technologist who will review it. Once it has been reviewed, the Music Technologist will delete the files from the computer's hard drive. So, if you have any reason to keep it past this review (if you wanted to do a different cut for the performer for example), note in the session folder name - DO NOT DELETE.

## LABELLING YOUR CD

When you've created your CD, label it with the name of concert and/or principal artist, the date of the concert, the venue, and indicate how many discs form the complete set. See the example below.



## CABLES AND CONNECTORS

In the professional audio world, there are three main types of analogue cable connection that you'll come across, and one of those isn't really professional, it's just common. These are described below –

### XLR



These are XLR connectors – Female on the left and Male on the right. Typically, the male side sends the signal and the female side receives it. This isn't always the case, but is usually, so knowing this should help make things quick to set up. These are the most common kinds of connection in the studio. They are used for virtually all microphones, most mixer to speaker connections, and to bridge long connections for instruments like guitars and synthesizers, with the use of a DI box. It's a very robust type of connection and the physically most secure type of connection in the studio. Use it wherever possible.

### ¼" Jack Plug/TRS/TS



Jack plugs come in two main flavours, both shown here – on the left a TS plug and on the right a TRS plug. TS stands for Tip, Sleeve and TRS stands for Tip, Ring, Sleeve. The former is a mono unbalanced connector and a TRS is either a stereo unbalanced or a mono balanced connector. The difference is that ring that goes between the tip of the plug to the sleeve (the long shank). It is used for fairly short cables runs only connecting many types of equipment to a PA or recording system. Guitars and keyboards use them. Balanced connections are intended to help combat interference problems. The male connector (the plug) tends to go on the cable at both ends and the female connector (the jack) tends to go on whatever hardware is being connected. The female end is usually only seen on cables as a way of linking multiple cables together. Although TRS cables can theoretically be used in the same way as XLR cables, they are rarely used with microphones or any other equipment that requires that phantom power be sent through the connector. The physical shape and arrangement of the connector would make this a fairly dangerous type of connection to send phantom power through. TS cables degrade audio signals over even moderately long cable lengths, so if needing to connect a TS connection to a PA system, it's important to use a DI box close to the instrument to convert the line level signal to a mic level, and at the same time change the connection from TS to XLR to carry that signal the rest of the way to the mixer or audio interface. The picture shows male connectors. The female connectors are always on the equipment you're connecting, regardless of whether you're sending or receiving a signal.

### RCA



This is the consumer cable of the bunch. It is used on most CD/DVD players and recorders and other consumer grade equipment. An audio only cable will typically come with two connectors on each end – a white or black connector and a red connector. Although it doesn't matter which colour gets connected to which side, it's always best to think that red is always right. Black or white is always left. A yellow connector is often used for video (the connection is called composite video). It can also be used to transmit audio. Since these cables only have two wires running through them, each cable can only transmit a mono unbalanced signal. That's why you typically see them in pairs. They should not be used for long distances. The end

shown in the picture is the male end. The female end is always on the equipment regardless of whether its sending or receiving the signal.

## Using a Turnaround or Gender Changer

The primary connection type we use in recording is called XLR. The history behind the name is complicated. It's on Wikipedia if you really want to know! Like other audio connectors, it has a male and female end, but unlike most connector types, the male end is used to SEND a signal, and the female end is used to RECEIVE a signal. There are exceptions to this, but it's good to get into the habit of remembering that male normally sends and female normally receives.



*XLR connectors – Female (receive) on the left, Male (send) on the right.*

Most XLR cables have a male and a female end for what should by now be obvious reasons – the signal is RECEIVED by the cable, flows through and then gets SENT on to somewhere else.

From this, we should be able to deduce that –

- Microphones have male connectors
- Mixer INPUTS have female connectors
- Mixer OUTPUTS have male connectors
- Speakers have female connectors

Our snakes that carry signals from one place to another and back again use the same concept, so at the mixer end of the snake, you'll have a whole ton of male connectors (typically 16-24) that connect to the inputs on the board, and a small number of female connectors (typically 4) that connect to the outputs on the board. At the stage end of the snake, there are a whole ton of female connectors that connect to microphones etc, and a small number of male connectors that connect to the speakers.

Sometimes we run into a problem that we need to return more than four signals to the stage – typically when we use more than two speakers plus two stage monitors. We get over this by changing one of the 24 male connectors to female at the mixer end and one of the 24 female connectors to male at the stage end. We do this using turnaround cables. These simply have the same connector gender on either end of a short cable. We have a few female to female and a few male to male XLR cables on the rack for this reason. If you can't find one, look in the top right drawer of the cabinets. Some are so short they don't fit on the rack.

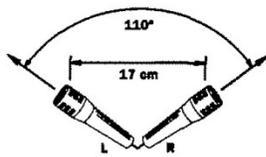
## One last point...

**All projects must be recorded on the CD projects log so that we know what's happening and what isn't.**

## EXTRA NOTES

### What about Suncor?

The process for recording in Suncor Hall is basically the same as in Cook, except that the microphones are not permanently suspended from the ceiling – they need to be set up before use. We also only use a single pair of microphones – there will be nothing equivalent to the catwalk mics that we use in Cook. Set up will typically take you an extra ten minutes to do because you're going to make the cable runs really tidy. You can grab a pair of microphones from the cabinet inside the door to Suncor Control Booth. They're in a black case. Grab two long XLR cables and a t-bar too. The t-bar is a black piece of metal, with two bolts sticking through it. You'll use this to hold two microphones on a single microphone stand. Use an ORTF microphone configuration (this is the same as the main mics in Cook Hall, and the diagram below).



ORTF microphone configuration. The distance and the angle are not an exact science. Somewhere around 90-110° will get you there, and the span from the edge of your thumb to the edge of your pinky will generally do for the distance between the mics. Perhaps of more significance, is placement in the middle of your performance set up, left to right.

# COMPLEX RECORDING SETUPS

So far, we have looked at simple four track recording. This is what we use for concert recording.

## MICROPHONE SET UP GUIDE

### PART 1 - INTRODUCTION

For serious recording of classical instruments, either as soloists or as ensembles, we typically choose a multi-microphone set up, using close microphones for attack and definition, and room microphones to add ambiance and richness to the sound. This guide will help you choose the right microphone set up for close miking, and give you some insight into your options for room miking.

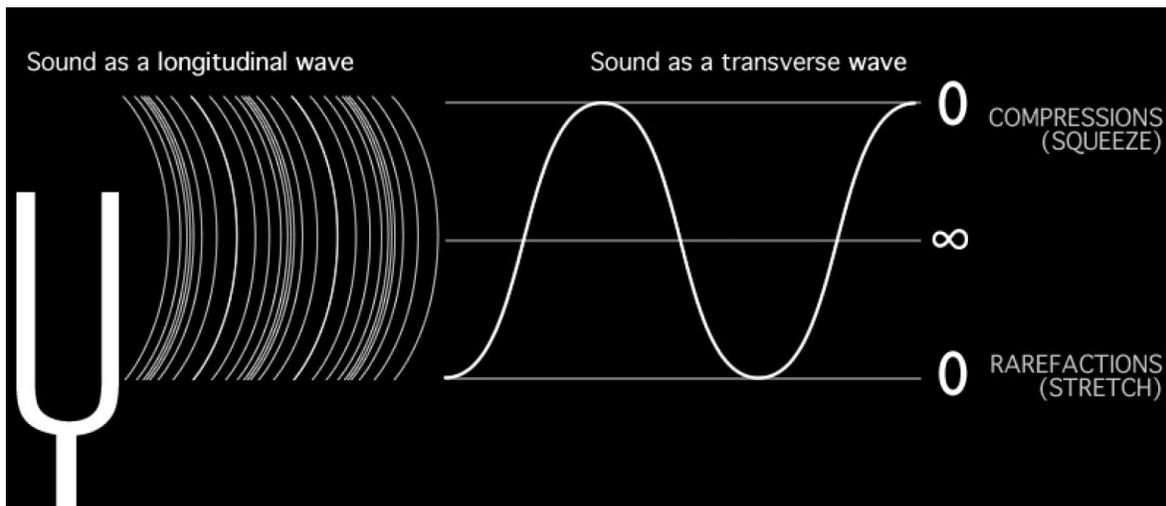
Before we get started, lets talk a little about sound...

#### What is Sound?

Sound is a mechanical wave caused by the vibration of an object that is transmitted through a solid, liquid or gas that is within the range of hearing.

There are two types of mechanical waves – transverse and longitudinal. Transverse waves oscillate perpendicular (90 degrees) to the direction of the energy and longitudinal waves oscillate in the same direction as the energy transfer. Sound is typically a longitudinal wave – if you think about how a loudspeaker works, it pushes and pulls the cone and the sound emits in the same direction.

In audio software, sound is represented as a transverse wave. The reason is because it's much easier to see what's going on visually in a transverse wave over a period of time. You might not see this now, but eventually you will.



Why do we care? Well, it helps you to understand what's happening with sound and why it is affected by objects in the way that it is. It also helps you to relate what you see on screen when you record to what is going on in the space that you're recording in. Sound represented by transverse waves have all the information that they'd have as longitudinal waves. It's just easier to read.

Sound vibrations are set up by musical instruments in different ways, and an understanding of where those waves are produced and how they're affected by the body of instruments can help you to determine good microphone placements when you're recording. If you know how sound is generated,

and propagated, you shouldn't need to learn specific microphone set up techniques for every instrument.

For classical music (and some other forms of music), a purest approach to recording is usually taken, whereby microphones capture the natural full sound of a musician or ensemble in its environment. This is in contrast to most pop recording where it is usually more desirable to record the driest sound that is not affected by room acoustics. This latter approach does allow for more definition or clarity and effects control which helps to define rhythmic and certain creative elements better, but it comes at the cost of some richness to the sound.

## Timbre

Timbre is what gives each musical instrument its unique character. The tonal qualities of musical instruments is important to listen for when you are setting up as microphone placement can dramatically alter what we hear in a recording, and in live sound reinforcement (A PA system). For the latter, we have to balance other environmental considerations such as feedback which can be introduced by our microphones picking up room reflections, and the sound quality is usually a compromise. But in recording, we can place microphones wherever we need to, to capture the most natural or pleasing sound without this issue. There are at least ten valid positions for miking up an acoustic grand piano. Each set up changes the tonal characteristics and the definition picked up by the microphones. Things that determine what kind of recorded sound you are aiming for include

- Traditions – positions and techniques other engineers before you have used for similar material.
- Repertoire – period and style will determine how much definition you need in your sound.
- Equipment – recording and musical equipment will determine where your microphones will capture the best sound.

There are three main areas that affect the timbre or tonal quality of a musical instrument when picked up by a microphone –

- The exciter – this is where the initial vibration is set up – the point of contact of the bow or finger with the strings of a violin or guitar, the lips of a brass instrument, reeds of a wind instrument or the beater of a drum or piano.
- The resonator – the body of the instrument.
- The environment – the room could be considered an extra body – its working in the same way as the resonator.

Microphones are usually placed where the best mix of these three elements happens -

Exciter	Brighter	More attack/definition	Less Echo/Ambience (Reverb)
Resonator			
Environment	Richer	Less attack/definition	More Echo/Ambience (Reverb)

However, there are other elements to take into consideration. Typically classical recordings emulate the ensemble sound from an audience perspective. So even though setting up microphones on a piano above the hammers would produce a bright but rich sound, and give a great stereo spread, it wouldn't produce a realistic emulation from the audience perspective. Great for pop, but not so for classical recording.

## Balance

To a large extent, we have an easy job to do with balancing the sound we capture. We expect our classical ensembles to perform acoustically, and they are trained to be well balanced within the ensemble. If we set up a pair of microphones in the approximate location of the audience, we should get a good representation of what the audience hears, especially if we use a microphone technique that mimics the human hearing mechanism. When we set up additional microphones however, we must determine the balance of a microphone within the ensemble, remembering that the ensemble includes the environment. We can use the mixer in our DAW to balance the various microphones that we have placed. Bleed of other instruments into microphones is to be expected and somewhat unavoidable, so listen to each microphone when it has been set up to make sure a distant instrument does not overpower a close one and that you don't have phasing issues (more about that in a moment). In other words, make sure the sound from each microphone gives a good balance of what it is supposed to be picking up. When using two microphones during a concert, the balance between the ensemble, the room and the audience is very important as it can't be changed in post production. In multi-channel recording, the balance is determined more during the mixing phase. However, the set up of microphones is just as important in order to make balancing possible in post production.

## Acoustics

So far, we've referred to Acoustics as Environment. They're interchangeable terms, but Acoustics is a more sound based term, so let's stick with that from now on. Any point within any room has unique acoustic properties. Even if you put strings on a trumpet, we wouldn't expect it to sound the same as a violin, and we shouldn't expect two different rooms to share the same sound either. Room surface materials and the positions of microphones within a room has a large impact on the sound we will get from that location. The instruments we are recording, in particular how loud they are, will also have an impact. Experience will guide us to the best location within a room to use as a starting point for microphone set ups. Our two primary recording rooms are Cook Hall and Suncor Energy Hall. They sound very different, and depending on the group, one might be better suited than the other. When setting up microphones it's important to listen to how reflections from the room interact with the direct sound from the instruments. Specifically we need to listen for two things –

- Reverberation: This is the room character. Reflections/echo. None is bad, some is good, too much is bad. Knowing where to put your microphones to get the right balance comes with experience, and getting advice from those that have done it before. If you have too much room sound, move your microphones closer to the instruments you're recording. If you have too little room sound, move the microphones further away. Changing the ensemble's position relative to the reflecting surfaces (walls typically), can also change the reverberation, though it's unlikely to significantly change the apparent acoustic space. For recording purposes, it's easy to put in additional microphones that allow us to capture the room sound and the close ensemble sound somewhat separately. This allows us to make the call on balance after the fact. For concerts though, it is usual to use as few microphones as possible so that we don't interfere with the audience experience so we need to get it right first time with a single pair of microphones.
- Phase Cancellation: This occurs when a signal is received from two places with a slight delay that interferes with the waveform. It can be caused by two microphones picking up the same signal, or by a single microphone picking up a signal from two places (usually direct, and bounced off a surface like the floor or a music stand). In both cases, it's the difference in delay of the two signals that causes phase cancellation. The result is a partial or complete cancellation of the wave form that at best can make instruments sound thin or unnatural, and at worst can

completely cancel out sounds. We check for phase cancellation in a couple of ways depending on the equipment we are using – we'll look at that hands on. Phase cancellation is corrected by moving microphones so that the delay no longer causes a cancellation of the signal. It may also require musicians to move.

## **Noise**

Noise is almost always bad. It can come from several sources, and sometimes you can do something about it, and other times you can't. So it's important to listen for noise and be able to determine whether or not you can do anything about it. The most common source of noise in a high quality system is environmental. Air handling units, air conditioning, heaters, mechanical equipment, even traffic, can cause noise and spoil a recording. Out of our two main recording spaces, Petro-Canada Hall has a worse noise problem than Cook, especially with traffic (and construction), and with the air conditioning. The lights can also create a lot of noise. The latter two, you can do something about. You can have the air conditioning turned off in the hall by phoning the Utilities Annex – the phone in the sound booth, and at my desk have the number programmed – just hit the down arrow on the phone until Utilities Annex is highlighted then hit Dial. Be sure to have it turned on after your session is over so that a relatively even temperature is maintained in the hall. The light noise can be prevented by avoiding using the house lights, ensuring as few lights as possible are lit, and that all those that are lit, are lit at 100%. The noise they create is a ringing sound. Be aware that if you change a lighting preset, it can cause the lights to click occasionally as they change temperature and expand/contract. So, if possible, set the lights even before you start setting up so you give them a chance to acclimatize to their new temperature. Traffic noise is a problem as is construction noise. This can't be avoided other than rescheduling a session late at night. Make a judgement as to whether it interferes with what you're doing and advise the artists if you think they should re-schedule. If in doubt, record a sample of the group, and have them listen to the result to see whether the noise is acceptable to them. In most cases, it is tolerable.

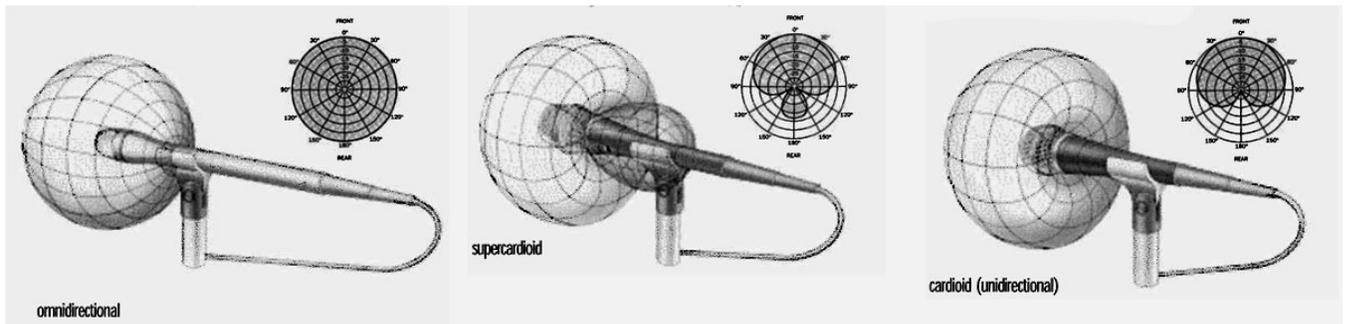
Other noise comes from the incorrect setting of gain structure within the recording system. It's caused by the equipment itself being used at settings it wasn't designed to be used at. We'll look at gain structure when we do our next set up, so we'll come back to this. Typically electronic noise sounds like white noise (like the ocean) but can also sound like a fax machine when connecting certain electronic devices to our set up!

## **Microphones – types, uses, precautions, technical requirements**

**Dynamic Microphones** – these are the most popular kinds of microphones. They are ideal for live sound reinforcement because of their resistance to feedback, their general ruggedness and their simple requirements to make them work! We have several types of dynamic microphones – Shure Beta 57, Shure SM57 and Audio Technica AT814. Their sensitivity and sound quality is not usually as good as that of other types of microphones (this is what makes them less prone to feedback), which makes them less well suited to recording situations. You can plug a dynamic microphone straight into any mixer in the building without any power requirements. Dynamic microphones work on the same principle as loudspeakers, only in reverse. In fact, you can use loud speakers as microphones, and in some instances, loudspeakers actually make really good microphones! Although I wouldn't recommend you play hackysack with them, a dynamic mic should typically survive a fall from 5-6 feet. You still need to be careful with them, but if disaster happens and they do take a tumble, they're not ready for the garbage yet.

**Condenser Microphones** – These are the most common microphones used for recording. We also have some we use for live sound, but they are typically recording microphones. Condenser mics are much more delicate than dynamic mics. They are likely to be killed if they are dropped from 4 or 5 feet, and since the ones we use range in cost from about \$500 to about \$1200, you need to handle them with extreme care and attention. There are numerous benefits to condenser mics, primarily an increased sensitivity (which leads to an overall higher quality), and they are available in polar patterns other than the stage standard, Cardioid (pronounced cardy-oid). Hey, polar what now? Okay, the pattern of the microphone diaphragm determines where the microphone will pick up the sound. Common types include Cardioid which picks up a 131 degree field in front of the microphone, to Omni which picks up a 360 degree field all around the microphone. Another common type is figure-8 which picks up on the front and back, but not on the sides. Why, you ask yourself? Well, sometimes its good to reject sound coming from certain directions, and different polar patterns help you to achieve this. Also, different patterns are required for different stereo miking techniques. We'll deal with that separately. For most recording purposes relating to classical recording, you can simply go by a rule of thumb that Cardioid is very universal and by far the most useful pattern – it can be used for practically every microphone use we have. Omni would be second most useful and would be particularly useful in picking up the room sound in addition to our instruments. It can produce a more open sound on things like piano, but beware that because it picks up all around, there are set up considerations that we must think about to avoid problems with noise and in a multi mic set up, with phase cancellation.

The following image doesn't show all the polar patterns, but is by far the best visualization I've ever seen of how polar patterns work in the real world.



...and this is the full range of patterns we can achieve using the microphones we have.

CHARACTERISTIC	OMNI-DIRECTIONAL	CARDIOID	SUPER-CARDIOID	HYPER-CARDIOID	BI-DIRECTIONAL
POLAR RESPONSE PATTERN					
COVERAGE ANGLE	360°	131°	115°	105°	90°
ANGLE OF MAXIMUM REJECTION (90° angle)	—	180°	126°	110°	90°
REAR REJECTION (relative to front)	0	∞	12 dB	6 dB	0
AMBIENT SOUND SENSITIVITY (relative to omni)	100%	33%	27%	25%	33%
DISTANCE FACTOR (relative to omni)	1	1.7	1.9	2	1.7

Condenser microphones require a power supply called phantom power in order to work. This typically comes from the audio interface or the mixer, and is often a button that says +48v or Phantom. The recording interfaces we use daily (the Digi 002R and the Digi Control 24) have phantom power turned on permanently, so you don't need to think about it. Incidentally, dynamic mics which don't need phantom power are not damaged when it is turned on, on our console. For live sound using the Soundcraft board, you need to set phantom on for any channel that you need it – we'll discuss that in more detail

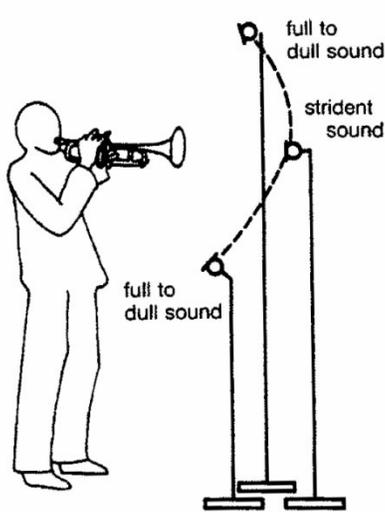
when we get to it. We have lots of different types of Condenser microphones – AKG C414, Neumann KM183, KM184 and Audio Technica AE3000 and AE3300. The AKG and Neumann microphones are high end, and are only used for recording. The Audio Technica mics are stage condensers and are used for live shows. They are designed to have better quality than the dynamic mics we have, but with some feedback resistance. The AKG C414 and the Audio Technica Mics are called Large Diaphragm Condensers (LDC's), while the Neumann microphones are Small Diaphragm Condensers (SDC). This simply refers to the size of the diaphragm inside the microphone that picks up the sound. Typically a LDC would have somewhat better sensitivity to low pitched sounds since its larger size lets it vibrate at slower speeds. In general, this means that they are typically better suited to low pitched instruments. In practice, all of the recording condenser mics we have are very well designed, and their use can be interchangeable.

**Other types of Microphone** – there are numerous other types of microphones, most commonly ribbon and PZM. We don't have any ribbon microphones which are incredibly delicate and have a very limited use. They are used in recording studios for solo musicians (really good on vocals and trumpets!). They generally have a figure-8 pick up pattern due to their physical design. We have a couple of PZM mics, but don't use them except in very rare situations. PZM's (Pressure Zone Microphones) are placed on a hard surface which acts as the microphone pickup. Although they have their uses they normally pick up too much noise from environment and from the surfaces they're placed on to really work... maybe we should experiment with them.... one thing to note with ribbon mics is that some can be damaged if phantom power is enabled on a channel that they are plugged into.

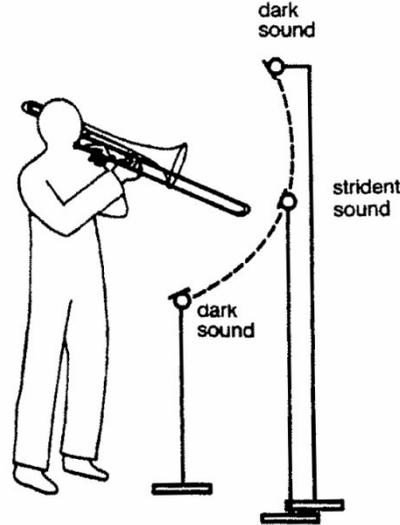
**Wireless Microphones** – we have one wireless microphone system that is available to use as a lapel mic or a headset mic. They are awful. Avoid wherever possible. If you do ever need to use them, they require batteries, so never offer them up on the fly. Both lapel mics and headset mics are condenser microphones (typically in a hypercardioid pattern), so while their design is intended to combat feedback somewhat, they are fairly prone to it because of the location they normally occupy on stage – especially the lapel mic which is right out in the open! They also have lots of handling noise. They are not suitable at all for recording purposes, and have very limited live sound use.

## PART 2 - CLOSE MIKING

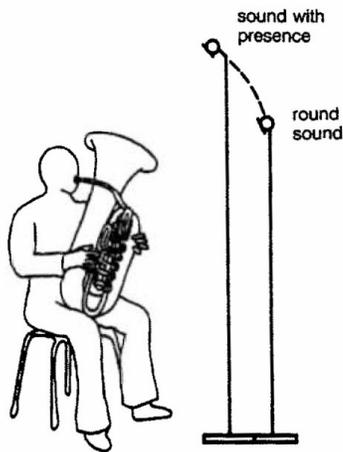
### Brass



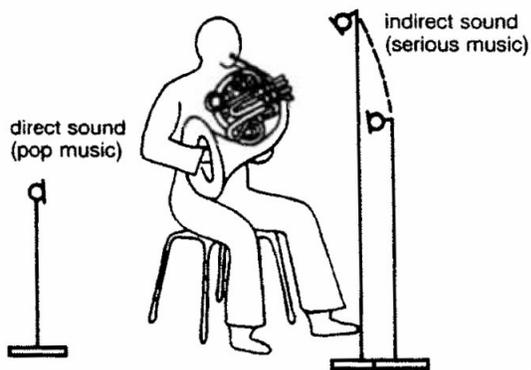
Trumpet



Trombone



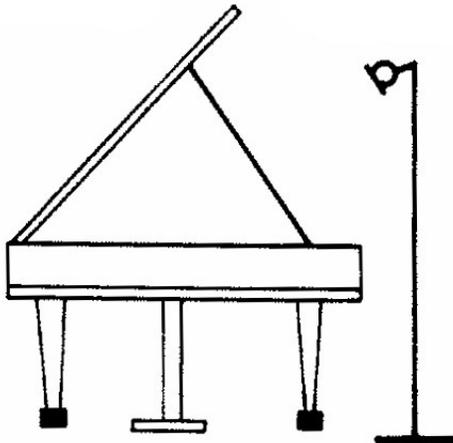
Tuba/Euphonium



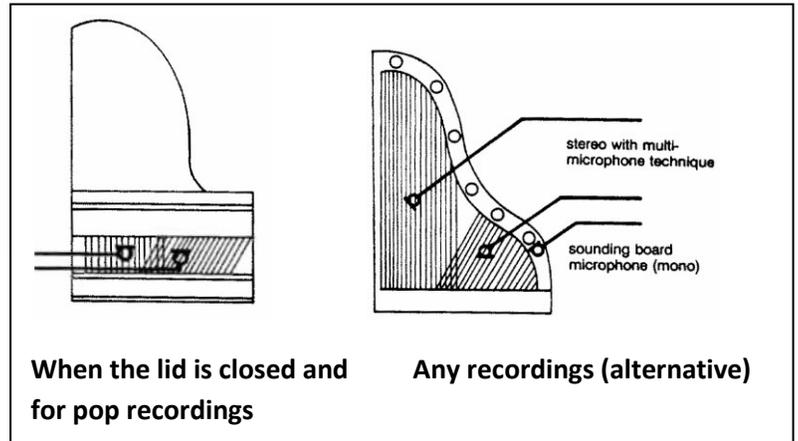
French Horn

The AKG C414 is a good mic to use on all brass. The Neumann KM184 would be good on trumpet, not so good on the rest. Use the direct (or as indicated in pictures – strident) position only for pop recordings and watch the levels – brass can get very loud! For classical, it has to be off axis for a richer, darker sound. Start at a distance of 3-4 feet from the instrument.

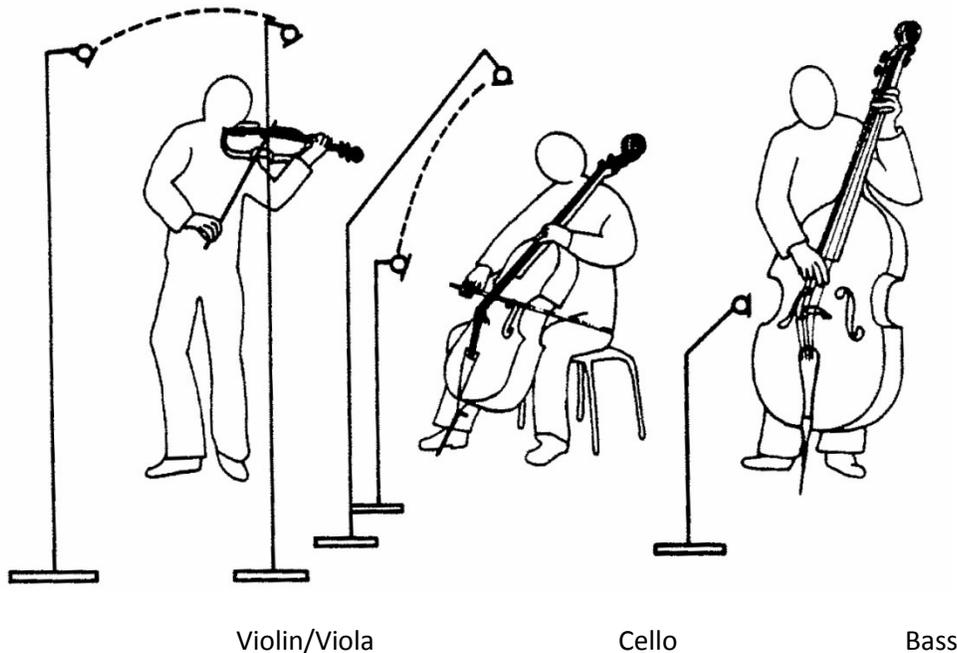
## Piano



First choice for classical recordings would be an ORTF pair, preferably Neumann KM184's on a t-bar placed in the piano nook (around where the lid catch is). Move towards the hammers for more attack or away for a darker sound. Start as close as the lip. Move out to reduce boominess or for a lighter sound.

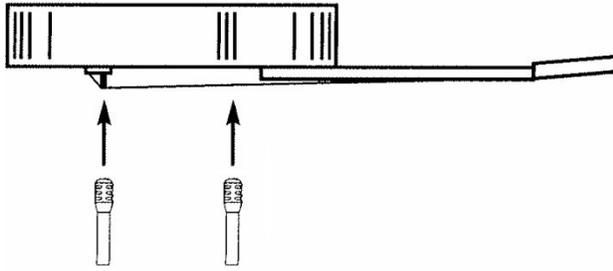


## Strings



As far as microphones go, the Neumann KM184 is your best bet on violin and viola, and the AKG C414 on cello and bass. Start 2-3 feet from the instrument. Move further away to avoid finger noise or for an airier sound or closer to maximize separation with other instruments.

## Guitar



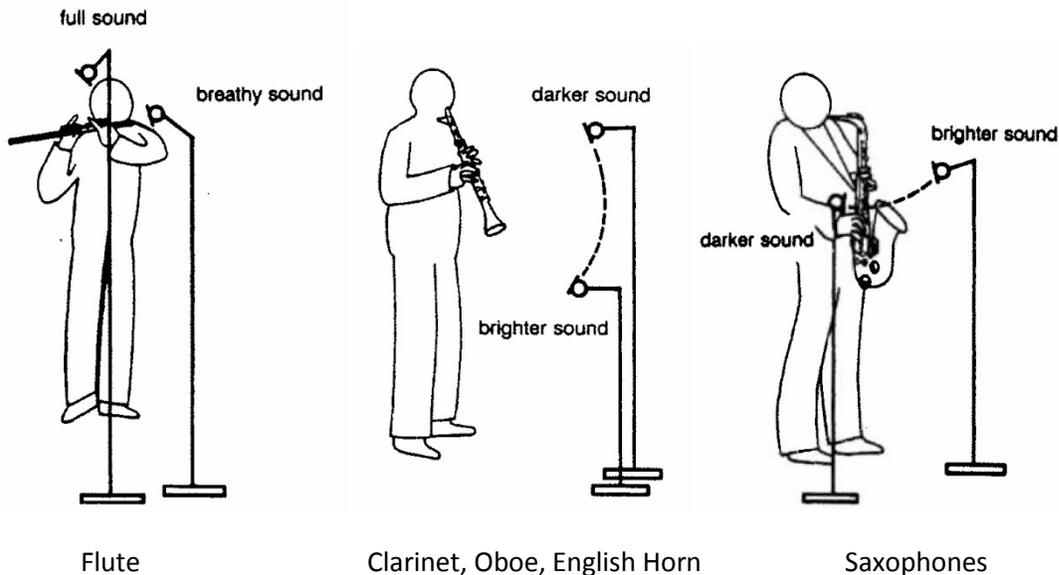
Guitar is treated differently to the other stringed instruments. As a solo instrument, it is typically miked in stereo as shown. Mics tend to go close to the instrument because it is so quiet. Just be aware of noise coming from the strings when miking really close. Although string noise is popular in acoustic pop, its not really in classical.

## Voice



Use AKG C414 in cardioid pattern, approx 4 feet in front of singer at chest height. Line up the mic so that the body blocks as much sound coming from the piano as possible. Normally singers won't use music on a stand. I wouldn't recommend you have them hold the music like in the picture. On a stand as far out of the way is good.

## Woodwinds



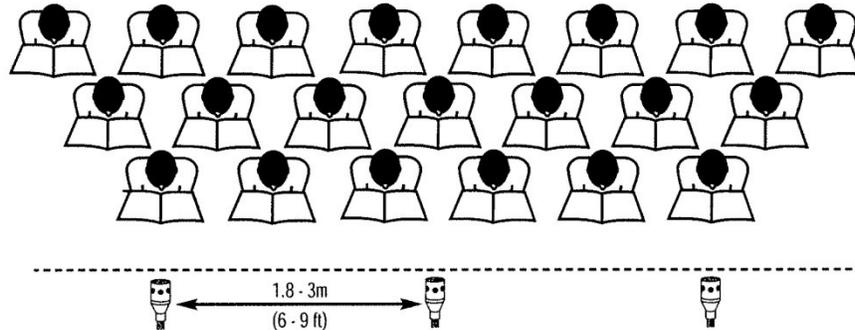
Neumann KM184 is the best choice for high woodwinds. Although the bassoonist will normally be seated, you can use the same principle of positioning as for clarinet. An AKG C414 might be a better microphone choice for baritone sax and bassoon. Start 2-3 feet from the instrument, except for a breathy flute sound which can be miked closer.

## Percussion

There are way too many variables for percussion to give it justice here. I would suggest using a “pop piano” style set up on bar instruments (marimba, xylophone etc.), and a close, slightly off axis position to the head/beater position for most other things. Percussion can be really hard to record well, especially triangle and tambourine.

## Ensembles

Its hard to predict how to record an ensemble. We don't have enough input channels to mic a large ensemble well, except for choirs, so you usually have to plan for those well in advance and make the most of what equipment is available. It will always be a compromise here. For choirs, the number of microphones you use depends on the size of the choir. Note the distance shown in the picture below – mics tend to get placed between 1.8m and 3m apart. Usually, the less mics you use the better, but you need to make sure the whole choir is covered, and well balancd.



## PART 3 - STEREO MIKING

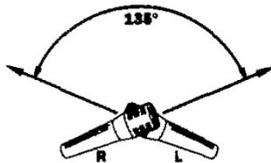
As mentioned in the introduction, in addition to using close microphones, we use room mics to pick up ambiance and a rich sound from the ensemble or soloist. In itself, the room sound doesn't have enough attack or definition to be used alone for critical applications (though we still use it for concert recordings). There are numerous stereo miking techniques that can be fun to try out and use. As long as you set up carefully, you can do little harm by trying any of them, but read the following carefully for precautions and recommendations.

Stereo miking techniques can be split into three groups – co-incident, near co-incident and spaced. What do we mean by co-incident? Basically, that the microphone capsules are in virtually the same space. The most common coincident technique is called X-Y – see the diagram below. Near co-incident is where the microphone capsules are close to each other but not intended to be in the same space. The most common near coincident technique is ORTF – see below. Spaced means that the microphones are deliberately distant from each other. The most common spaced technique is actually called... spaced pair.

### COINCIDENT TECHNIQUES

Coincident techniques are typically used to provide maximum mono compatibility. Useful if the material will be played on the radio, but less important for other mediums. It provides an accurate image, but tends to be a little on the narrow side which may prevent your recording from sounding airy and rich. The exception is Mid-Side (see below) which has a very variable image width. Because the microphones are very close together, it may be hard to pick up a large ensemble accurately using this technique.

#### X-Y



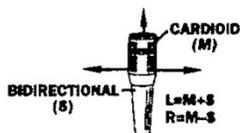
This technique uses two cardioid microphones with the capsules crossed at 135 degrees. You don't need to be there with a protractor... anywhere from 90 to 135 degrees should be fine. Neumann KM184's work best for this set up.

#### Blumlein (or stereosonic)



Blumlein looks very similar to X-Y – the biggest difference is the type of microphone you use. It must be a bi-directional (figure-8) microphone. For us, that means you can only use AKG C414, set to figure-8. Get as close to a 90 degree angle as possible.

#### Mid-Side (M-S)



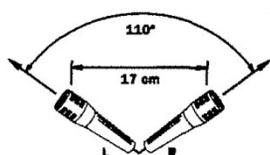
Hard to see from this picture, but the Mid-Side technique typically uses two microphones – one cardioid pointing forward, and a bi-directional (figure-8) microphone pointing to the sides. This picture shows a purpose built MS microphone with everything built into one casing, but you can do it just as well with two mics. You also need to decode the signal inside of Pro-tools when using this technique. It's fairly simple – you set up one channel for the cardioid and two channels for the

figure-8. The figure eight channels are panned hard left and hard right. The cardioid channel panned centre. Then you invert the phase of the right side figure-8 channel. Now when you ride the faders for the figure-8 channels (together), you increase and decrease the width of the stereo field. Very cool. Works nicely as a stereo microphone technique for classical guitar and would also be good to give spaciousness for anything unaccompanied. The only matched microphones we can use for this are AKG C414's.

## NEAR COINCIDENT TECHNIQUES

The Near Coincident techniques provide a wider stereo image that give a sense of air and richness to the sound that you don't get with coincident techniques. It's less compatible with mono (more prone to phase cancellation). Like the coincident techniques, because the capsules are close together, this might not be the best way to capture a large ensemble.

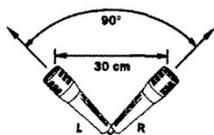
### ORTF



ORTF is a technique that was first used by the French broadcasting organization (**Office de Radiodiffusion-Télévision Française**). It is our most commonly used stereo miking technique today. For this, we use a pair of cardioids microphones with a specific spacing (17cm) between the capsules, and at a specific angle (110 degrees). You can estimate 17cm by using the

distance between the tips of your pinky and thumb when you span your hand as widely as possible. Yes, this is approximate since it depends on the size of your hands, but saves carrying a tape measure around with you. Typically we use Neumann KM184's for this technique, but we can use AKG C414's too.

### NOS



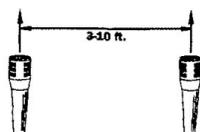
NOS was first used by the Dutch broadcasting foundation (**Nederlandse Omroep Stichting**). Although it looks very similar to ORTF in the picture, note the significantly different space between the capsules – 30cm instead of 17cm. The angle is slightly less too – 90 degrees. Like ORTF, we use a pair of cardioid

microphones for this. Neumann KM184's are the natural choice.

## SPACED TECHNIQUES

Spaced techniques provide the widest image of all and are least mono compatible. They are prone to phasing issues so need to be checked carefully for problems. They are the best choice for large ensembles, and for the largest feeling of spaciousness.

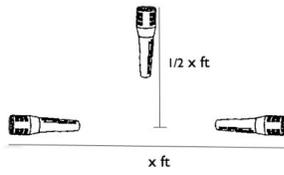
### Spaced Pair



The spaced pair is the most common spaced technique. Although the picture indicates the distance between microphones should be 3-10ft, we typically place them wider apart than this, and use the technique as much to record the space itself as the ensemble. In Petro-Canada Hall, I typically use a spaced pair 20ft apart with good results.

## Decca Tree

The Decca tree uses three microphones (sometimes more – typically an odd number), and is common on film stages and large ensembles such as orchestras. There are no fixed measurements for this technique, but as seen in the image, typically will consist of two widely spaced microphones, and a third placed half the distance in front of the side mics as the side mics are widely spaced. For this technique, omni mics are used. Even though omni mics pick up all around them, face them toward the side walls for the side mics and to the front for the middle mic. Groups that use this technique often have special large brackets on a single stand for this to work. We don't have anything like this and would put them on three separate stands. Typical distance between the side mics might be six feet (making the centre mic go 3 ft out front).



**The most common techniques we would use here are ORTF, spaced pair, and possibly Mid-Side.**

### IMAGE ACKNOWLEDGMENTS

**Shure Microphone Guide** - polar patterns, image of guitar and choir microphone set ups

**Tonmeister Technology (Michael Dickreiter)** - all other instrument microphone setup positions

**Shure Microphone Guide** - stereo microphone technique images, except Decca Tree

## TIPS FOR MANUAL SET UPS

Knowing how to set up from scratch will help you troubleshoot any problem, and give you maximum flexibility in custom set ups. Understanding some of the issues you need to watch for when creating a custom set up will help you to avoid recording issues that can't be resolved in post-processing.

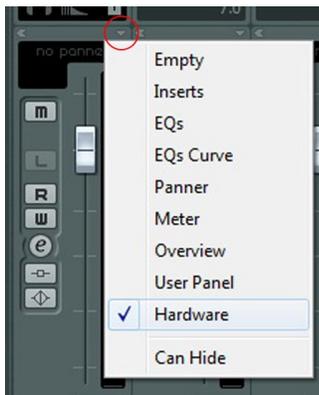
The big difference between the interface in Cook booth and the primary interface in PC Hall, is that the one in PC Hall has built in hardware DSP. Even if you're not planning to use the effects, its vital that Nuendo is correctly set up to avoid recording these DSP effects in with the microphone signal.

Custom set up might include any combination of stereo and mono inputs that give you the best flexibility and ease of editing of any project. Multiple outputs allow you to set up a Cue Mix that lets the performers hear what they've recorded (or if required, hear what they're recording as they're doing it).

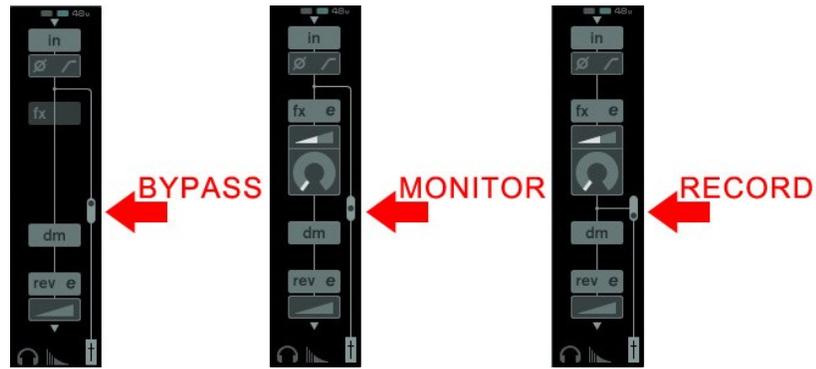
So, everything we're talking about here comes under the broad term of routing. Lets get started.

As mentioned above, there's one big difference between the interface in Cook Hall booth (a Steinberg MR816x) and the primary one in PC Hall (a Steinberg MR816csx). This is hardware DSP. The chances of you wanting to use hardware DSP are very small indeed. They're more for pop singers who we record at close range without lots of room ambience than for classical recording. I'll go through how to set up a Cue Mix later which you could use to provide the performer with a mix including DSP, but to keep things simple, we'll just concentrate on the settings within Nuendo that prevent you from inadvertently recording DSP.

In your mix window, you'll find the inputs on the left, tracks in the middle, and outputs on the right. The DSP settings are part of the input channels. In order to show them, click the small arrow on an input channel to reveal the options dialog. Choose Hardware.



Above the mix faders, you should now see the hardware routing that will look like one of the three images below.



Note the small rounded rectangle switch that sets the routing to one of three settings. Click on the switch to change from one setting to another. Looking closer at the whole image, you'll notice that it shows you how the DSP signal will be routed.

**BYPASS** – you'll want this setting 99.9% of the time. This will mean that DSP is completely bypassed. No one will ever hear it.

**MONITOR** – you'll want this setting if you set up a Cue Mix and your performer wants to hear themselves with reverb and/or compression effects.

**RECORD** – you never want this setting. This will add any DSP to the audio input and it will be mixed and recorded with the audio input. This cannot be undone, and once you record it, there's no way to remove it. That's why you never want to use it.

Check this setting for each input channel you have set up. In theory you'll be able to hear the difference, at least between the BYPASS and the other two settings. In practice, you could miss it, so always check the settings visually.

While on the topic of routing, let's take a closer look at the input and output connections. This applies to both Nuendo and Cubase.

We route inputs and outputs using busses. A buss becomes the container for any physical input or output on the audio interface. They are globally set up using the VST Connections window – Devices > VST Connections.

Tabs along the top of the VST connections window help you navigate between inputs, outputs and a whole bunch of other busses. They are irrelevant – the only ones important to you at this point are inputs and outputs.