EXPANDING THE CHURCH AUDIO ENGINEER'S TOOLKIT

by Stephen Compton
PhD Candidate, MA, BA(Hons), Dip. Sound Engineering

Connection, Collaboration, Experience and Creativity

A 'Quick-Guide' Training Supplement

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This document was originally compiled as part of my Masters study. This supplement is designed to assist the training of church audio engineers and act as a quick-guide reference that addresses important audio topics and practise. It is not a comprehensive all-you-need-to-know manual.

A fully referenced companion Live Audio Handbook is available from classaudio.co.nz and addresses hearing management, sound levels, monitoring SPL and more.

I trust this document provides a useful resource that helps in Expanding the Church Audio Engineer's Toolkit.

"He that has an ear let him hear..."

"Blessed are your ... ears because they hear." Matthew 13.16

"Each of you should use whatever gift you have received to serve others,
as faithful stewards of God's grace in its various forms". 1Peter 4:10

"Whatever you do, work at it with all your heart, as working for the Lord" ... Col 3:23

"The greatest among you will be your servant." Mathew 23.11

"A faithful man will abound with blessings". Proverbs 28:20 .

"And let us consider how we may spur one another on toward love and good deeds".
Hebrews 10:24

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The live-sound 'experience' is powerful, connecting feelings, memories and people: physically, emotionally, socially and spiritually.

Value, Purpose and Connection

Within our churches, the audio engineer stands in a pivotal and rarely understood role as a creative multitasker and problem solver.

From the stage, the musicians and presenter may provide the source audio, but it is the audio engineer that makes crucial decisions that weigh up all contributing sonic influences, and the opinions and expectations of others.

The church audio engineer is a gatekeeper, directing the flow, by which the 'experience' is shared. Their role requires more than making things louder. Instead they help facilitate an atmosphere whereby participants can focus on the aims of the service without distraction, feeling comfortable to express, and receive input into, their faith.

Although positive acknowledgement for such a crucial position may not be as forthcoming as it could be, the position can provide great personal satisfaction and be an expression of the audio engineer’s worship and service.

Purpose and connection drives each volunteer, leader, musician, operator and congregation member’s actions, thoughts and communications.

Church audio production is a collaborationn that supports the pastor, the worship leader, and the musicians to accomplish a common goal while taking into consideration the needs and desires of the congregation, while also balancing the practise of live-audio production.

- **Many aspects of sound are within the sound operator’s control but there are aspects that are not.**
- **Stakeholders each contribute to the success of a church and the church service.**
- **Common aims and goals, relationships and responsibility are key to successful collaborations.**

Common Aims and Individual Responsibility

Leaders

- Hold an understanding of how stakeholders, building acoustics, equipment, training and processes impact eachother.
- Provide sincere acknowledgement of volunteers.
- Manage staff and expectations.
- Manage the budget.
- Equip stakeholders with resources.
- Encourage mentorship, training, ownership and relationship.
- Direct the flow of information: service planning, preparation, rosters.
- Receive ongoing personal developement.
- Build connection with other stakeholders.
- Develop a culture of excellence.

Musicians

- Develop skills and practice.
- Are in tune and makes appropriate tonal, dynamic and rhythmic choices.
- Are cohesive and blend with others, fitting within musical arrangements.
- Are responsible for personal onstage volume.
- Receive ongoing personal developement.
- Embrace and learn the equipment used.
- Understand the necessity of connection with other stakeholders.
- Able to express their faith genuinely in front of others
- Develop a personal faith
- Respect authority

Sound Operators

- Are prepared and calm,
- Are attentive and diligent.
- Maintain good personal hearing.
- Learn problem solving techniques.
- Receive ongoing personal and skill developement.
- Embrace and learn the equipment used.
- Connect with other stakeholders.
- Aim for seamless practise that does not draw focus to the mechanics of sound reinforcement.
- Understand the impact their role has in the focus, flow and seamlessness of the service.
- Allow their audio engineering to be one expression of their faith.

Congregation

- Actively participate in fulfilling the church aims.
- Understand the limitations of volunteers.
- Develop an attitude of encouragement and kindness above critique.
- Develop and maintain relationship and connection with other stakeholders.

onstage noise

Qualities of a Good Church Audio Engineer

- They enjoy working with sound and with sound equipment finding personal satisfaction in the task.
- They can musically ‘feel’ the emotion in music.
- They recognise their audio engineering is an expression of their personal worship.
- They maintain good personal hearing health.
- They develop problem solving skills.
- They develop patience and calmness under pressure.
- They are a negotiator, teacher and multitasker.
- They can function without the need for acknowledgement.
- They balance expectations.
- They balance audio in a musical way.
- They are diligent, vigilant and conscientious.
- They are couragious, thick skinned and resilient.
- They embrace and learn their technology.

Factors Influencing Sound

- Building acoustics.
- Stage noise.
- Musician and vocalist skill.
- Skill of the operator.
- Tuning.
- Instrumentation and orchestration.
- Rhythm.
- Equipment.
- Collaboration, relationship and cohesiveness between participants.
- Each participant fulfilling their role responsibilities.
- Expectation.

As with learning an instrument, audio engineers can make noise relatively easily. The more that critical listening and audio engineer skills are studied and practised, the better the resulting sound can be.

"Everything that is said and done relates to successfully fulfilling that purpose and ultimately for the Glory of God."
- Chris Huff

Balanced Plugs and Cable

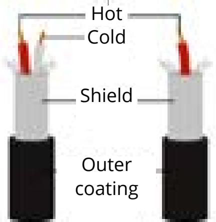
- XLR = Extra Long Run 250 metres +.
- For microphone-level signals.
- For line-level sources.
- To transport stage audio sources to the console, and the console to the amplifiers.

Unbalanced Plugs and Cable

- Must be shorter than 6m long or radio interference. or other induced noise may mix with the signal.
- Used for electric, acoustic and bass guitars, keyboards, DIs, AV, iPods over short cables etc.

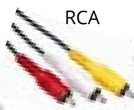
6.35mm TRS Jack (tip, ring, sleeve)

XLR (hot, cold, shield)



Balanced Unbalanced

6.35mm TS Jack (tip, sleeve)



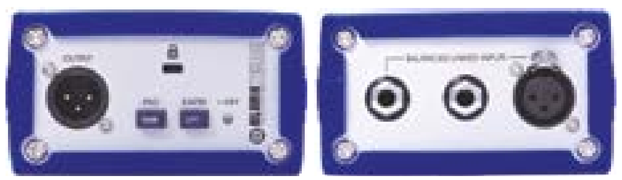
RCA

As stereo jacks these 6.35mm and 3.5mm jacks are unbalanced

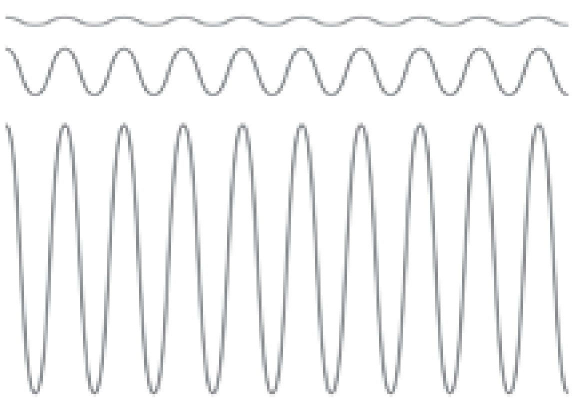
Stereo Jack (tip-left, ring-right, sleeve-shield)

DI - Direct Input/Direct Injection/Direct Interface

- Converts an unbalanced signal to balanced and:



- Reduces earthing issues,
- Pads audio (reduces the level 20 - 30dB),
- Provides a 'through' port to **send** the signal to the mixing console **AND** to an amplifier,
- **Active DIs** require phantom power (+48v) supplied from a battery or the mixing console
- **Passive DIs** do not require phantom power.



Mic level - The tiny voltage produced from the vibrating microphone diaphragm requires amplification from a mixing console 'gain' pre-amp to increase the signal to a useable level.

Line/Instrument level is generally a higher signal level than a microphone level. The console may need to provide only a small amount of gain, or none at all, to be at a useable level. Jack inputs prepare a console for line-level input. Consoles that only have XLR inputs may have a PAD button or MIC/LINE INPUT selector button to prepare the console for line-level input.

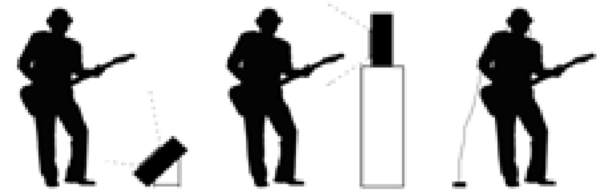
Amplifier level - Cables from an amplifier are unbalanced, but carry a large signal voltage so do not generally suffer noticeable induced noise.

Speakon connector for amplifier level signals.

CAt5, Cat5E, Cat6 ethernet cable with RJ45 connector, used to pass digital multichannel audio or network signals.

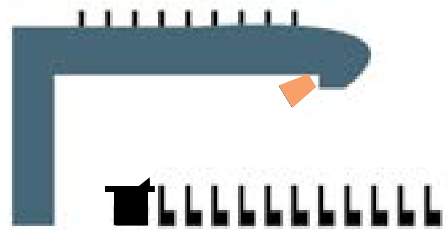
Reducing Stage Noise

- Musicians can lower their personal acoustic level.
- Reduce foldback volume.
- Face instrument amplifiers to the musician or position the amplifiers off-stage.
- Provide the musicians and singers with in-ear monitors (IEMs) which also aid communication, and allow for using pre-recorded backing with click tracks.
- Install screens around acoustic drums or use electric drums.
- Install acoustic treatment to reduce and control audio reflections.



Acoustic drums with screens [and lid]

Electric drums

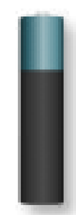


Mix Postion

Mixing from under a balcony, in a booth, being too far from the direct sound of the speakers, or away from where most of the audience are, compromises how an operator hears a performance and in turn affects their audio choices.

- The optimum mix position within a venue, is where the operator can hear sound similar to what most of the audience hears.
- Acoustic treatment, speaker placement and system design may improve the consistency of sound.
- Wireless control of digital consoles allows the engineer to walk around and check how the mix sounds throuighout the venue and make adjustments or accomodate the .
differences in sound response throughout a venue.

Battery Usage:

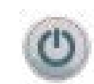


Outside of a 'service', test how long radio microphone or in-ear monitor batteries are expected to last, by:

1. Reading the battery voltage using a multimeter and taking note of the type of battery and starting voltage.
2. Insert the batteries into the device and switch 'ON'. Note how long the time is before the microphone or IEM no longer transmits, receives, or turns OFF.
3. When testing, some batteries require a rest/recovery period before their voltage can accurately be measured again.

"The alkaline batteries I use, provide a minimum of 8 hours use. After a single use, I know that, given a recovery period, and that the batteries measure above 1.4v, I can rely on getting a minimum of 4.5 hours more power. The rechargeable batteries I use that measure deliver 5 hours of continuous use when they measure at a full charge of 1.3v." SC

Computer Audio:



- Computer audio should preferrably be passed through a USB, firewire, thunderbolt interface rather than directly from the built-in computert audio output.
- When using the built-in computer audio output keep cable length <6 m to the front of house mixer, or use a DI.
- When a computer is connected to a second screen via VGA, a buzz may appear on the audio line require a DI or interface to 'clean' the signal.

Service Preparation and Setup

Careful preparation reduces stress, allows time to solve any issues and leaves the audio engineer to focus on the creative aspects of the mix.

Before the Musicians Arrive

1. PURPOSE
Understand your role in relation to church aims, and with other people.

2. TURN EQUIPMENT ON IN THE CORRECT ORDER
a. Desk/effects ON.
b. Amps/speakers ON.

3. CHANNEL ALLOCATION
a. Determine the best channel allocation for quick, easy access, positioning related items close to each other.
b. Label the mixing desk clearly.
c. Label vocal mics left to right from the engineer's perspective
d. Link the stereo channels if both the channels will be treated in a similar way e.g. drum overheads, keyboards, AV or playback device.

4. PLUGGING IN AND PREPARING MIXER
a. MUTE ALL input channels.
b. Plug in all microphones and instruments.
c. Leave the channel faders down. **Only** put the Group masters and Aux master knob/faders to 0 (U).
d. Put LR (FoH) at 0 (U) or cautious at -10dB.
e. Put HPF on all but the bass, kick and maybe keys.
f. Put phantom power on DIs and condensers.
g. Select PREFADE for all foldback sends except for track playback and effect sends which will be POSTFADE.
h. Group each channel to the Group Masters e.g. band to 1&2, vocals to 3&4, and playback tracks straight to LR.
i. Adjust pan/balance to direct the audio left or right between the stereo group selected.

Making Noise and Final Checks

5. TEST SPEAKERS AND THE ROOM
a. Play a 'clean' sounding track and listen to each front of house speakers to check they sounding as they should, without distortion, and aimed correctly.
b. Do the same for foldbacks. This process also checks the **sends** are **sending** signals where they should.
c. Play a track you know well AND in the style of the service to hear the influence of the room, temperature and humidity. Walk around the room to hear how even the sound coverage is, and listen for any variations in volume and tone.

6. CHECK ROUTING with a music player and desk mic
a. Check all channel routing to groups and FoH.
b. Check all desk routing to foldbacks.
c. Check all desk routing to effects.
d. Check the batteries in radio mics have enough charge for the rehearsal, and service.
e. Check the radio mics are working and that their radio frequencies are not interfering with other radios.
f. Check lapel/headset cable/plug for damage.
g. Test AV and playback devices.
h. Line-check all channels if possible.

Preparing and Labeling a Mixing Desk - an example:

ALL channels should be in a logical order and labelled in such that each can be accessed quickly. If the console has layers, put the items requiring the most active adjustments on the top layer, and then carefully choose which can be on the second layer. A drumkit may be a good option to put on a second layer. Use colour codes, text or pictures to visually separate sources. Group (or use VCAs or DCAs) 'like' channels and label for quick identification and access.

SETTING GAIN STRUCTURE:

Every piece of equipment adds some degree of noise.
- **Optimising gains will reduce the amount of unwanted noise, provide good levels of audio for use with foldback sends, and deliver the best signal to and from microphones, instruments, AV equipment, amplifiers and speakers.**
- **Much of the gain structure process can be achieved while the musicians are setting up, making noise and preparing for rehearsal.**

OUT of one - IN to another

OUT mouth - IN mic
OUT mic - IN mixer
OUT mixer - IN amp
OUT amp - IN speaker

OUT mixer send - IN effects unit
OUT effects unit - IN mixer channel

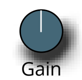
OUT mixer - IN system processor
OUT system processor - IN amp
OUT amp - IN speakers

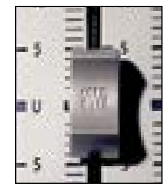
KEEP CHANNELS WITH DIS AND CONDENSERS MUTED WHEN INSERTING OR REMOVING THE PLUGS FROM MIXER CHANNELS OR SNAKE. NOT DOING SO WILL CAUSE LOUD POPS THROUGH THE PA AND CAN CAUSE DAMAGE

Musician Soundcheck

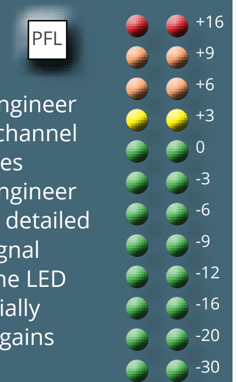
7. PREPARE EACH CHANNEL FIRST IN SILENCE
a. Fader down, Gain low *. Make sure instrument/mics are plugged in
b. Press PFL/SOLO.
c. **Adjust gain** while musican/singer plays/sings at performance volume. Increase the gain on individual channels until the the signal in the meters for:
- **Analogue console: reaches 0dB. Occasionally go to +3, max +6.**
- **Digital console: max of -12dB.**

d. Make sure there are no 'clip's and do not increase the gain knob past 3 o'clock.





8. PREPARE EACH CHANNEL FOR THE PA
a. Unmute the audio channel.
b. Raise the fader WITH CARE to performance level. **Not past 0dB (U).**
c. EQ appropriately. 1 fix, 2 blend, 3 enhance.
d. Turn the appropriate **Aux** knob to **Send** the channel audio to Foldback (PREFADE for all foldback sends except for track playback).
e. **REPEAT FOR ALL INPUT CHANNELS**



PFL/Solo:
- allows the engineer to listen to a channel on headphones
- allows the engineer to get a more detailed view of the signal levels using the LED meters especially when setting gains

* If the channel allocations do not change much from week to week or if presets are used on a digital console, a few of the above steps may be bypassed.

The minimum front-of house volume should be just above the acoustic on-stage volume. During rehearsal lower the FoH volume to check the acoustic level. If this is still too loud the band and/ or foldback level will need to be lowered.

The Rehearsal

The rehearsal may be the only opportunity to walk around and hear what the musicians and the congregation are hearing in various parts of the building.

9. MIX and MONITOR
a. Sculpt a mix using volume, pan, EQ
b. Monitor the input meters and the output meters to check there are no 'clips' exceeding 0dB. **If the level 'clips':**
- Reduce the gain on the channel that is 'clipping'.
- Reduce channel fader levels feeding a group that may be 'clipping'.

c. Walk onstage and check the individual ONSTAGE foldbacks, and in different parts of the room to check how mix sounds.

10. CHECK THE LEVELS THE FADERS ARE AT.
No **channel faders** should be over 0 (U). Some will be at zero and some lower. Close to zero will allow greater fine adjustments. Adjust the speaker amplifier volume so the Master fader is also close to 0 (U).

Before the Service Starts

11. SAFETY AND AESTHETICS
Check all cables are not a tripping hazard and are visually tidy.

12. PRE-SERVICE AMBIENCE
Play appropriate pre-service music at a level people can talk but also creates the desired expectation and 'atmosphere'.

13. REST EARS
Take a break away from noise for a few minutes.

14. JUST BEFORE SERVICE START
Progressively increase the volume of the pre-service music so when the musicians start to play, the volume isn't too much of a shock to the congregation.

15. DURING THE SERVICE

- **Be prepared:** Effective preperation makes the mixing process easier, more creative and with more focus.
- **Be attentive:** Be ready for any changes or fixes required.
- **Aim for seamlessness:** like changing gears in a car aim for smooth transitions that cause no distractions.
- **Anticipate:** Prepare for what comes next with no missed audio. Keep a consistent volume and tone, with fingers on faders and knobs ready when microphones are passed between people, or when more than one microphone is used.
- **Observe and respond:** Watch the meters to avoid distortion. NOTE: Changes in channel GAIN also affect foldback levels. During the service only change the GAIN to fix 'clipping'.
- **Observe and respond** to SPL meter readings to assure participants of safe levels.
- **Observe and respond** to musicians and singers to ensure they hear what they need to.
- **Observe and respond** to congregational involvement. Your contribution will be evident in congregation responsiveness. An audio engineer should adapt their mix to each specific audience.
- **Make musical audio choices** and be aware that the choices you make impact each stakeholder.
- **Make intelligible:** Speech and singer intelligibility is vital to a good mix.
- **Allow dynamics** within the music. Have moments when the music is quiet and subtle, and times when it is louder.
- **The Presenter:** The presenter/MC should sound as natural sounding as possible: warm, no accented sibilance, and with minimal compression. The congregation should 'lean' in and not be 'pushed back' in their seats. Generally NZers do not like to be shouted at.

Note: If a sound operator can obtain the appropriate volume within one word, listeners may believe the operator has had the correct volume from the outset.

After the Service

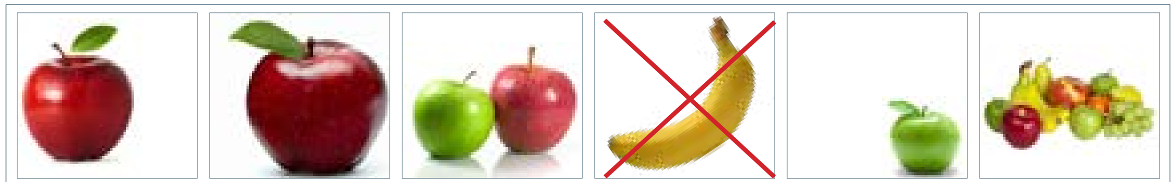
16. Turn OFF the PA system:
a. Amplifiers/speakers OFF.
b. Everything else: desk etc OFF.
c. Put the batteries on charge.
d. Tidy cables, mics, console and workspace ready for next engineer.
e. Check with the musicians and leaders on how the sound impacted them.
f. Provide positive encouragement as a team member.

POWER, BUZZES AND HUMS:

- Make sure all audio power is on the same mains phase.
- When running audio on two or more mains phases a transformer may remove potential noise issues.
- The electrics in some building's add noise to audio that can be difficult to eliminate.
- Audio cables that cross AC adaptors e.g. keyboard transformers, may induce noise on a line.
- Using a good DI with an earth-lift button may reduce problem noises.
- Swap XLR cables or snake channel allocation to test where a connection, or other potential fault causing noise may lie, or test with a cable/plug tester.

When two mic cables are connected to two identical microphones or DIs with the same gain structure on both channels, and one channel is 3 - 6 dB louder than the other .... one of the wires on pin 2 or 3 may have sepereated from the plug terminal.

Cable and plug testers



Referencing: Comparing 'apples with apples'

To mix well, learn to listen! Know what to aim for!
Learn how each source should sound, how to make them fit and blend with other sounds, and what is appropriate for a given event.

- Compare one sound with another.
- Critically listen to individual vocal and instrument volume and tone comparisons within commercial recordings, especially with songs peformed at your church.
- Learn to match tones, stereo position, dynamic range, effects and how each sound interacts and fits with other sounds.
- Establish what is most important, be it a lyric, the message, song leader, instrumentalist, or rhythm, and make sure the most important things in your mix are heard clearly.

Tips for critical listening. Take note of:

- What is the loudest and clearest sound source (normally vocals).
- How does the vocal sound? How 'warm' or 'muddy' is it? How much 'body' is in it? How much can the 's' and 't' be heard? Can you hear breath?
- Throughout the song, take note of any changes in volume with each sound source (dynamics).
- Take note of the blend of vocal harmonies and the volume and tone of each of the parts.
- Which instruments or vocals sound closer or further away, to the left or to the right or in the centre.
- How does an acoustic guitar sound when just with a vocal? How different does an acoustic guitar sound when it is part of a band?
- How does a piano sound in a classical type scenario vs a pop scenario or as part of a band? A classical piano will generally be 'warmer' with less high-frequency content than a pop-style piano recording.
- Observe how each element in a drum kit sounds taking note of the tone and the relative volume of each.
- When a bass guitar and a kick drum play together what is the tone of each?
- What effects are on sound sources to make them sound like they are in a particular space.

Make a playlist of songs that include:

- Tracks that are used at your church (commercial recordings).
- Tracks you personally know very well.
- 'Clean' tracks to determine if there is any distortion in the speakers.
- Tracks that have a wide stereo image.
- Tracks that contain deep full bass.
- Tracks that contain crisp highs without being too harsh.
- Tracks that are known to be mixed very well.
- It can be useful to record short audio clips with "left" and "right", "in phase" and "out of phase" to test signals are going to where they should and interacting properly in mono zones.
- It is always useful to have playlists ready for quieter moments in a service when there isn't a band or if the band are otherwise occupied. It is also useful to have a selection of appropriate pre and post-service tracks should they be required.

Microphones

Finding the right microphone

Considered mic choice and placement are a key to capturing the best source audio. Different microphones will have have different frequency responses so trial microphones to find which is best suited for a particular voice or instrument.

CONSIDERATIONS: SPL of instrument, type of microphone and its frequency response, sensitivity, ruggedness, price, polar pattern, appearance, placement and treatment.

Dynamic	Condenser
Close to source	Can be further from the source
Better isolation	More sensitive
Harder wearing	Less rugged
Proximity effect	Brighter tone
Less Expensive	More expensive
Quiet handling	Noisey handling
Less prone to feedback	More prone to feedback
Phantom power (+48v) OFF *(won't damage mic if phantom is ON)*	Phantom power (+48v) ON *(won't work if phantom is OFF)*
FOR ANYTHING CLOSE OR WHEN ISOLATION IS REQUIRED e.g. vocals, guitar cabinets, percussion, brass.	FOR WHEN NOT CLOSE TO SOURCE OR NEEDING DETAIL e.g. choir, lectern, orchestra, piano, strings, winds, cymbals and percussion, recording.

Other Specialist Microphones

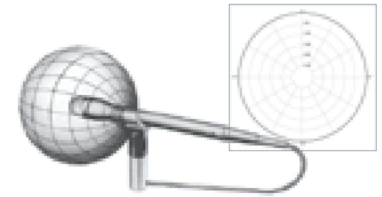
- For high SPL (Sound Pressure Level) e.g. Kick drum.
- Visually discreet e.g. PZM, clip-on.
- Ribbon microphones have a flatter response curve so are more accurate, though fragile and comparitavely expensive. **Do not use phantom power on standard ribbon microphones.**
- Shotgun microphones are useful for capturing sources at a relatively long distance away directly in front of the microphone.

Polar Patterns

Microphone polar patterns provide a system to not only determine where a microphone DOES collect sound, but also where a microphone DOES NOT collect sound.

OMNI

- Picks up all directions

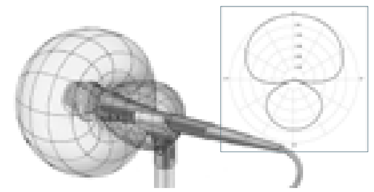


CARDIOID

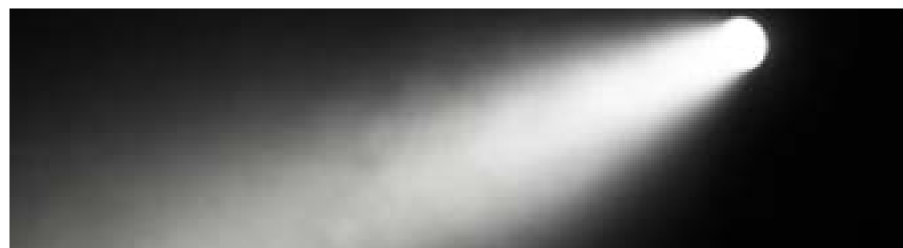
- Picks up front and sides
- Rejects behind
- Good if positioned directly in front of a foldback or noisy instrument

HYPERCARDIOID

- Picks up front and a little behind
- Rejects 45 º behind
- Good if foldback or noisy instrument is at a 45 º axis



Other patterns include figure of 8, shotgun, semi-circular and super cardioid



Radio Microphones:

- Radio microphones and wireless in-ear systems must be within legal frequency ranges.
- Interference from other radio frequencies is possible as radio microphones and wireless in-ears become commonplace. A less expensive radio microphone may be able to select between a small number of frequencies to find a 'clean' band. More expensive radio microphones may have the ability to scan the radio spectrum and suggest 'clean' frequency options, syncing to various devices.
- Physically touching the radio microphone aerial may create an audible noise.

Other Microphone Considerations

MUTE channels when plugging or unplugging items ESPECIALLY CONDENSERS and DIs .

'Plosives are large bursts of low-frequency energy caused by a voclist's 'P' or 'B' .
To fix:
- Adjust how the microphone sits in relation to the mouth. Some experimentation may be required, as not everyone expresses air in the same direction.
- Wind socks can reduce 'plosives and are often essential outdoors in windy conditions.

The Proximity Effect is a low-mid frequency boost and occurs when in close 'proximity' to a cardioid or hypercardioid dynamic microphone. This boost can be favourable. The effect can be reduced with an EQ or positioning the microphone a little further away.

Limit Feedback:
- Close-mic the source.
- Encourage a vocalist to hold their mic close to their mouth and sing confidently.
- Microphones should be faced away from speakers.
- Use an appropriate polar pattern according to where foldbacks and other sound sources are positioned onstage.
- Lower the foldback levels.
- Use a feedback destoyer.
- EQ out problem frequencies using a multiband EQ.

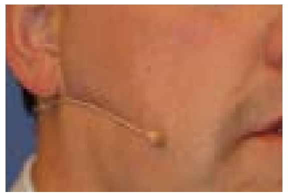
Finding the right placement for a microphone

LISTEN to each instrument and amplifier by carefully moving your ear to various positions around the instrument. Put the microphone at the exact place that your ear determines is the sound you want.

CONSIDER the isolation, bleed and capturing the desired sound of the instrument.
Good choice of mic and placement reduce the work required to get a good sound at the mixing console

Placement - Single microphones on a single sound source

A FEW EXAMPLES:



VOCALISTS:
- Choose a microphone that has a tonal character that flatters the vocalist's natural tone.
- Dynamic microphones generally have a mid-low boost when a vocalist is close, and the sound is 'thinner' the further they are away from it.
- Choose either a cardioid or hypercardioid to best reject sound from the position of the nearest foldback speaker in relation to the vocalist.
- Encourage a vocalist to use a microphone in such a way as to reduce 'plosives, or use a wind sock.
- Not every vocalist sings directly forward.The microphone may best be placed to one side of the mouth.

HEADSET MIC:
- Place close enough to get plenty of level from the mouth.
- Place far enough away to avoid breath/wind noises or 'plosives.
- Sit gently on the cheek as being closer to the skin will pick up more low-mid frequencies.

DISCREET: Clamping to a music stand or instrument can be visually appealing and save floor space.

FLUTE:

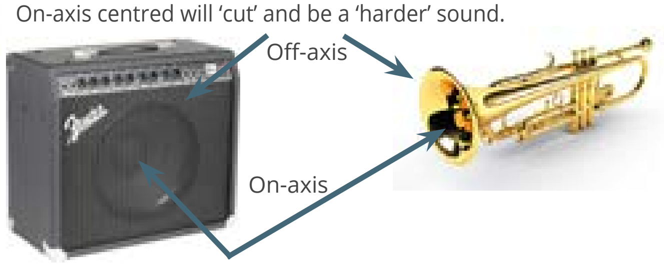
- Close-mic using a dynamic microphone. The most common mic position is closer to the mouthpiece, although further down the flute is where the air escapes. Close-mics will pick up more breath sounds. When recording use a condenser mic above, allowing a small distance.

ACOUSTIC GUITAR:
- If not using a acoustic guitar pickup (which is preferable in live situations), a dynamic microphone placed close will pick up more of the instrument and less of other instruments than a condenser, which is more commonly used when recording.
- A microphone near the sound hole may not necessarily produce the best sound. Slightly towards the neck may give a better balance between fullness and clarity.

VIOLIN:

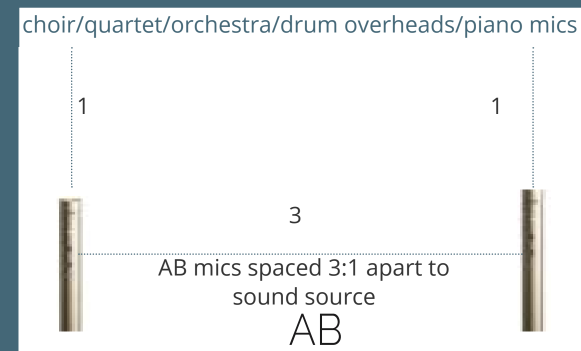
- Clip-on condenser close-mics will have a 'grainier' sound of the bow on strings.
- A dynamic mic will provide better isolation.
- Condenser microphones further away will have smoother sound but will pick up other sound sources as well.
- 'The Band' pickup system is a contact option that stretches around the instrument.

GUITAR AMPS AND BRASS:
Off-axis placement will produce a warmer sound with less 'cut'.
On-axis centred will 'cut' and be a 'harder' sound.

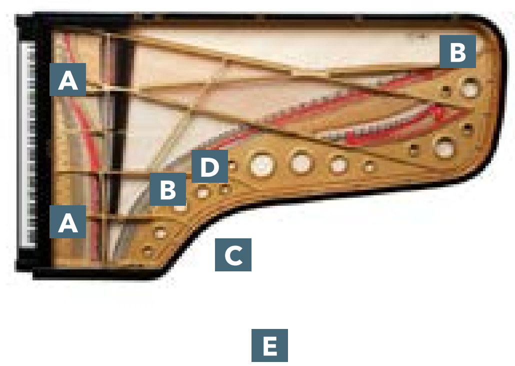


Placement - Multiple microphones on a single sound source

Two microphones on the same source may give a fuller, wider stereo image but their interaction may also have a detrimental impact and make the sound 'smaller'. By adjusting the distance between the mics and/or to the source or pressing the phase (Ø) button on one input may make an improvement. AB and XY stereo techniques are useful for choir/quartet/orchestra or drum overhead microphones.



PIANO:
Listen closely at various positions around and in the piano as the pianist plays.
Place the microphones where your ear determines has the sound you want.



Starting Points :
A Condensers AB 3:1 spacing - clear and defined notes, some pedal noise possible. Can be used with lid open or closed.
B Rich grainy bass, and general coverage. Lid open. Level or below lip.
C Single or XY condensers. Lid open. When recording in a average room acoustic.
D Single microphone or XY for general coverage. Lid open. Vertically half way between lip and lid.
E Classical sound. Warmer with less cut, more acoustic, and less isolation.

DRUMS:
With 1 mic: 1 x kick OR overhead.
With 2 mics: 1 x kick, 1 x overhead.
With 3 mics: 1 x kick, 2 x overhead or 1 x kick, 1 x overhead, 1 x snare.

Kick: The further the mic is into the hole the more slap of the beater is heard. Use a mic that can withstand high sound pressure levels (SPL). Two mics can be used in the kick. One to pick up the thump and another inside for the slap e.g. Sennheiser 901 and 902.
Snare: A mic on the top picks up the body of the snare, a mic on the bottom picks up more of the chain/crack. If two mics are used then the signals can interact well to get the desired sound, BUT due to destructive interference may combine to a 'thinner' sound. Press Ø on one of the channels to see if the sound becomes fuller. If it does, then leave Ø in.
Overheads: Use condensers in XY or AB placement or a single condenser over the centre of the kit.

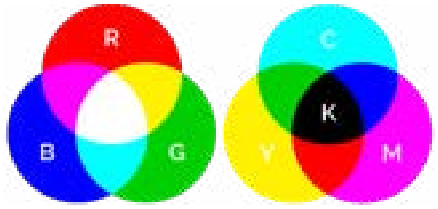
Close mic'ing of each element improves the isolation of sounds.
Using more microphones does not necessarily produce a better result!

The complex colours we see are a combination of simple colours. They are a mix of simple colours. For example:

RGB - Red Green and Blue or

CMYK - Cyan, Magenta, Yellow and Black.

The sounds we hear are also a combination of simple parts.

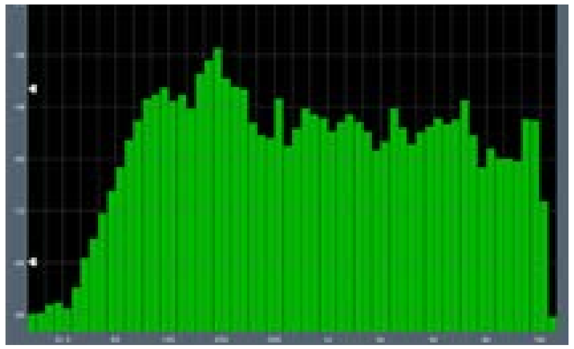


EQ-Equalisation-Tonal control

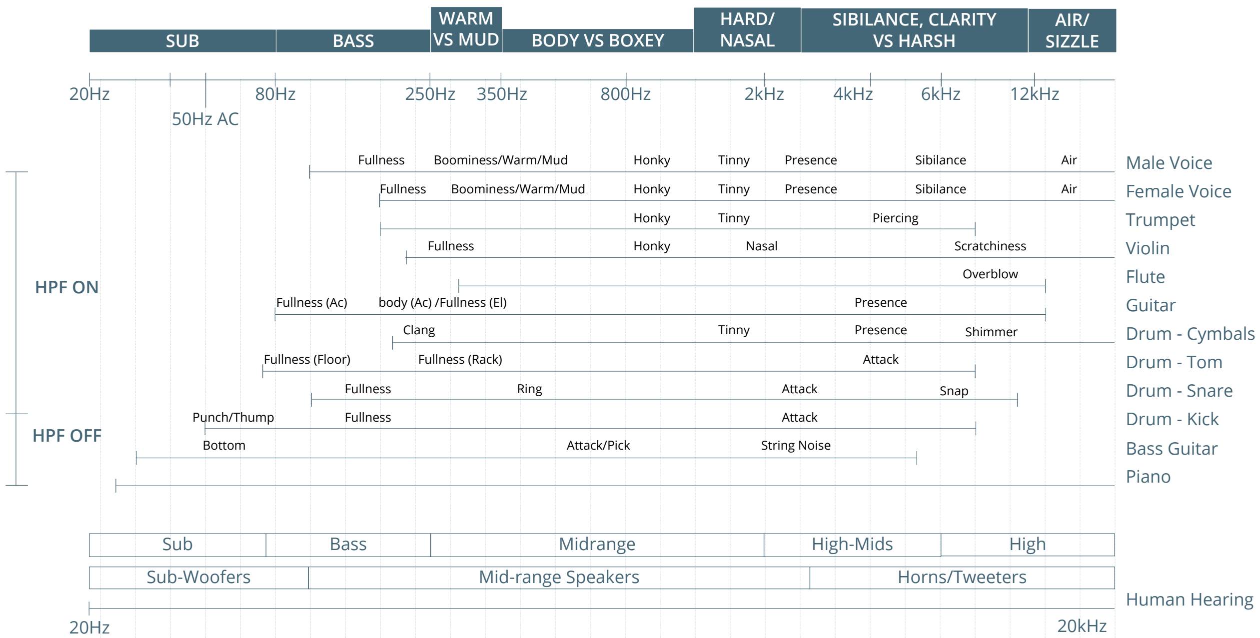
The complex sounds we hear are a combination of simple air pressure wobbles, which we recognise as sound wave frequencies (simple sine waves).

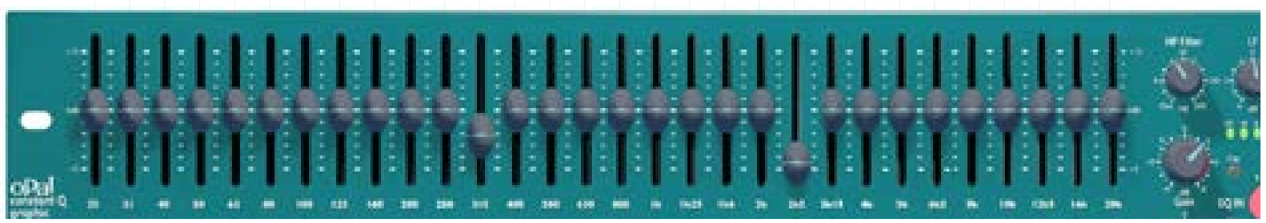
Equalisation is a process of reducing or increasing the levels of these simple sound wave frequencies that make up each 'complex' sound.

EQ allows the user to fix, enhance, spatialise or assist in blending one sound with another sound.

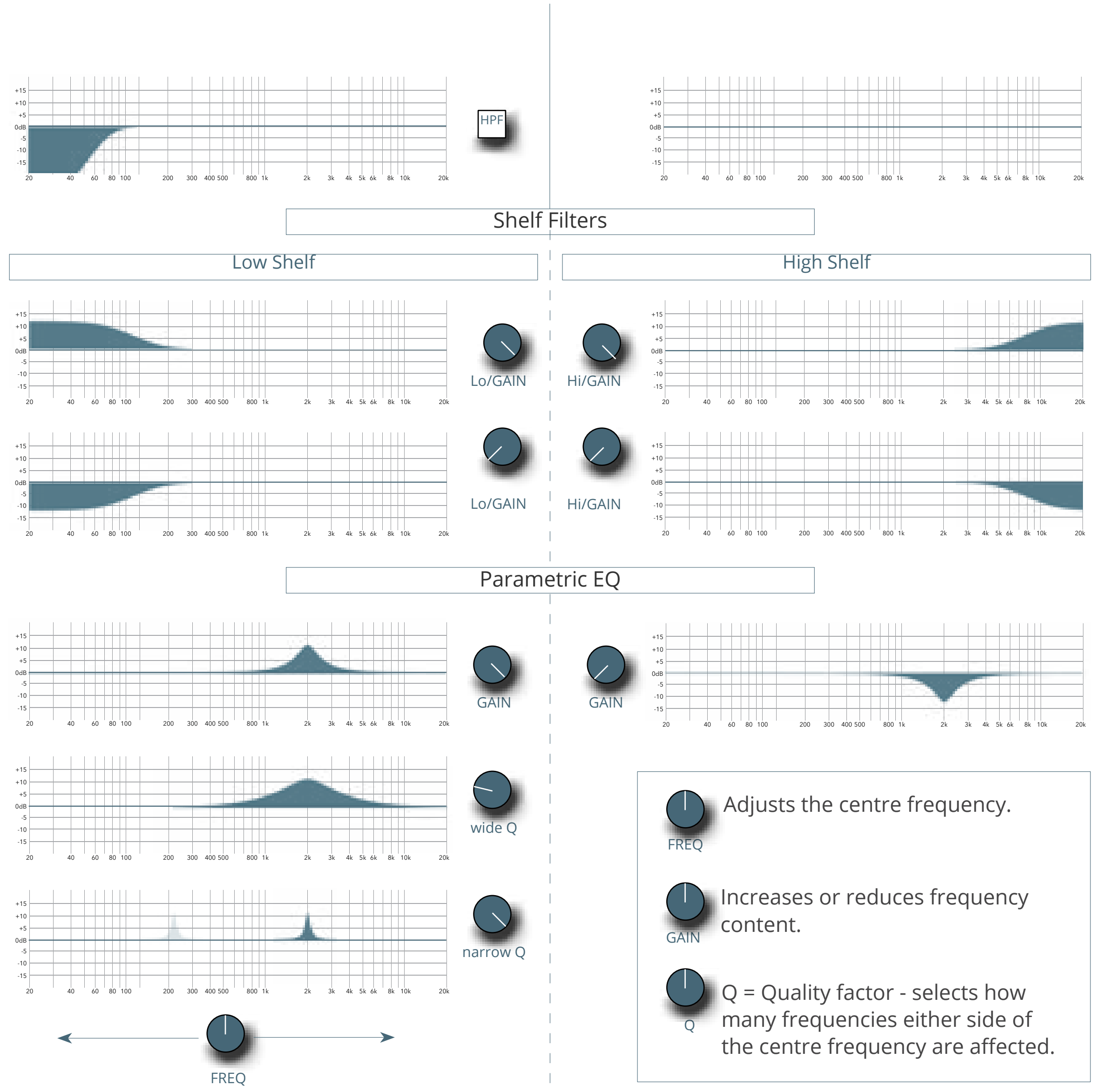


Above is a visual representation (FFT) of the many simple frequencies present in a song at a single point in time.

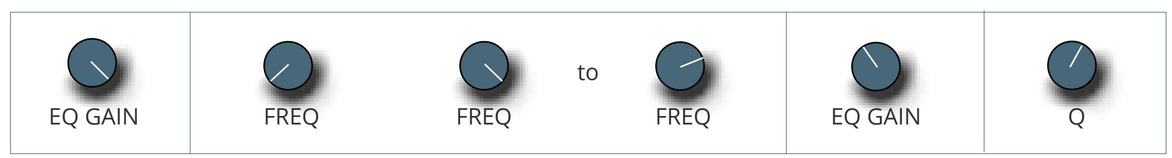




31-band EQ used for 'notching out' narrow problem frequencies eg feedback.



'SWEEP' to FIND frequencies - ONLY DO THIS DURING REHEARSAL!!!

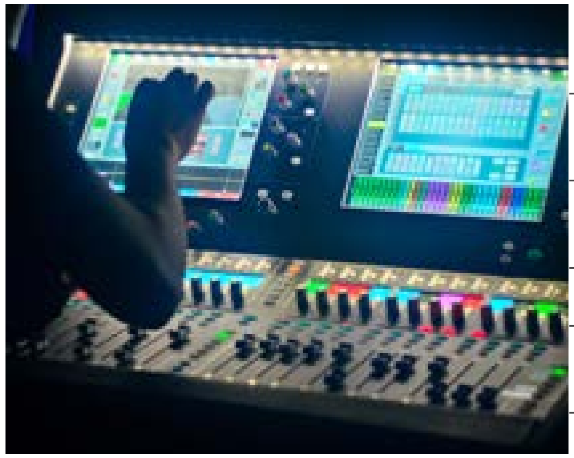


1. Increase the frequency band gain.

2. **'Sweep'** the frequency knob till you hear the desired frequency accented

3. Reduce the frequency band gain until the chosen freq is at the right volume level.

4. Adjust the Q so that the EQ changes balance being natural sounding while also targeting the intended frequencies.



When using an EQ

- Listen to lots of music, particularly commercial tracks of the songs performed in your church, and listen to the tone of each sound.
- Find a combination of words that describe each sound source's tone e.g. warm, muddy, boxey, bright, harsh, nasal, sibilant.
- Use a High Pass Filter (HPF) on all but kick, bass guitar, AV and maybe the keyboards.
- By reducing high-frequencies and lowering the volume, a sound source will be 'pushed back' into the mix.
- If singers have trouble with timing, reduce the level of sibilant frequencies (4-6kHz) will make multiple occurences of consonants less noticeable.
- Some instruments will be EQ'd differently depending on the other instruments playing at the same time. e.g. An acoustic guitar with a vocal: The guitar can be 'full' sounding and EQ'd so that each word of the vocal is heard clearly. In a band situation the bass guitar and keys provide most of the 'low-end' and so the acoustic guitar can be less full, focusing on the mid-highs and highs to provide the 'cut' and rhythmic strumming sounds.
- A good start when EQ'ing instruments and voices, is to make them sound as they do acoustically, although the kick in a contemporary drum kit may favour being fuller, defined and louder.
- It is generally better to cut frequencies than add.
- Choosing appropriate microphones and considered placement is better than relying on an EQ to 'fix'.
- Consider using an EQ to scoop out frequencies on instruments that compete within similar frequency spectrums.

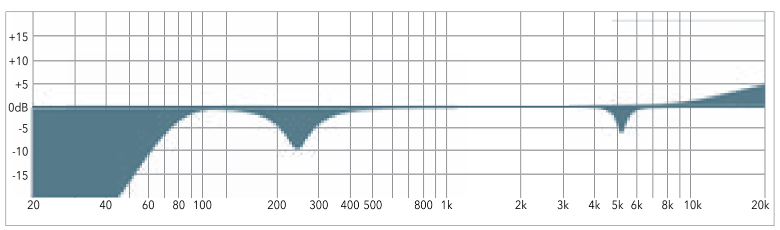
DON'T JUST EQ SOUND SOURCES IN ISOLATION. WHEN MUSICIANS MAKE MUSIC, EACH SOUND SOURCE WILL 'BLEED' INTO OTHER MICS AND HAVE AN IMPACT ON THE VOLUME AND TONE OF EACH ELEMENT

The process of Equalising: **1 Fix, 2 Fit, 3 Enhance**

1. Press the HPF button if the frequency content of an instrument or vocal is not lower than 80Hz. This will reduce the 'bleed' of other low-frequency sounds into other microphones. HPF is normally ON for all but the kick drum, bass guitar and sometimes on keyboards.
2. Make sure that what is 'most important' is clearest e.g. Lead vocal. Instruments supporting these sources may need to be slightly duller and at a lower volume sonically 'pushes them back' in the mix making the vocal feel more 'forward'.
3. Listen for the warmth/mud 250Hz-350Hz area. Use the low-mid frequency sweep knob along with gain to balance warmth without being too 'thick' and 'muddy' sounding.
4. Listen for clarity around 4 - 6kHz. Balance the high-mid sibilant frequencies so the 's' or 't' don't stick out too much but has enough to maintain clarity.
5. Make further adjustments while other sound sources are playing to encourage the 'blend'.
6. During the rehearsal if there is a frequency area that doesn't sound right compared to your 'reference', sweep, find and fix.

Not every sound should be crisp and clear. Likewise not every low-frequency sound should have a low frequency boost

Right: An example of what an EQ curve may look like with an HPF ON, lowering the muddy and sibilant frequencies and adding a little 'air/sizzle'.



Feedback

- FoH speakers should be forward of the stage and in front of all microphones.
- Face microphones away from speakers.
- Use microphones with suitable polar patterns to reject sound from the direction of the speakers.
- Lower the volume of the speakers.
- Use a HPF on all but bass, kick and possibly keyboards.
- Notch out potential problem frequencies using a 31-band EQ or a feedback destroyer.
- DO NOT put a lapel, headset or lectern mic through their own foldback speaker.
- Compression can make feedback occur easier.
- Condenser microphones are more sensitive than dynamics and therefore have a greater potential for feedback to occur.

Fixing Feedback During a Performance:

1. **STOP THE FEEDBACK QUICKLY** by **MUTING** the offending **CHANNEL**. Muting the channel will also mute the foldback if the foldback is fed from the same mixer.
2. Establish quickly the cause of the feedback to determine whether it is the result of a temporary action by performer or from another cause.

High-Frequency Feedback - More commonly caused by microphone through foldback.
Low-frequency feedback - From front of house **OR** foldback.

If the feedback is not caused by a temporary performer action:

3. Lower the foldback GAIN a bit, then lift the volume of the channel to front of house cautiously. Return foldback slowly but not quite to the previous level and listen for hints of feedback starting.
4. Feedback frequencies can be 'notched out' using a graphic EQ or a parametric EQ.
5. As quickly and as seamlessly possible, allow the performance to continue.

KEEP
CALM
FIX IT
AND
CARRY
ON

Good preparation limits potential issues but may not eliminate them. After an event where feedback has been an issue, review PA speaker positions to determine if these positions can be improved to reduce the liklihood of feedback while maintaining the coverage required

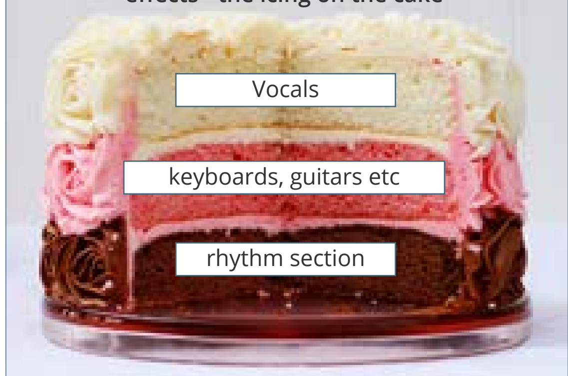
The key to a good mix is:

- knowing what sound and levels to aim for and what is appropriate for your church.
- to make what is **most important heard** albeit the dialogue, lyric, performer, lead instrument, lead vocalist **AT the appropriate volume** for the particular occasion.
- Allow an **appropriate dynamic** range and **tone** that makes the vehicle of sound invisible.
- Treat each audio source in such a way that they fit with other sources musically, and at a volume and tone that fits within the culture of the event.

The Mix

- Mixing skilled musicians and singers is much easier than mixing less-skilled musicians and singers.
- The musician and singer's tuning, timing and rhythmic issues will affect the engineer's ability to make the mix 'sit' well.
- A skilled audio engineer can make a poor sound system sound ok while a poorly skilled operator can make even a good system sound poor.
- Every song or presenter requires different tonal and dynamic treatment.
- The final mix must fulfill the aims of the church and not biased by personal taste eg making a guitar heavy mix.
- In a church environment, the most important channels are the vocals. The words need to be heard clearly.
- Make the lead vocal slighty louder and clearer than the backing vocals.
- MC and presenter's mics should be natural sounding and not overly loud. Let the congregation 'lean' their ears in, instead of being conscious of amplification and 'pushed back' in their seats. NZers do not like to be shouted at.
- Bass frequencies are important for vocal pitching. Sub speakers make a 'full' mix easier and less harsh. When a mix is 'full', it is often perceived as being less loud.
- Appropriate microphone choice and placement will reduce the need to 'fix' the sound with EQ.
- Some congregation members may find a song mixed differently to recordings, distracting. Others may find a different and well-executed mix refreshing.
- A well-mixed band with skilled musicians can be louder without causing complaints than a quieter, novice mix with less-skilled musicians.
- Not every instrument has to be at the same volume. Some can be subtle, and some require changes in volume.
- Allow the congregation's ears to 'breathe'. Some songs should be quieter than others.

effects - the icing on the cake



The Layer Cake Analogy:

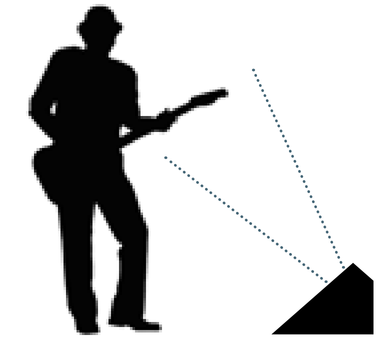
TOP Layer: The vocals are at the top of the 'mix' clearly above the supporting instruments.
MIDDLE Layer: Adding colour and flavour, other instruments build on the rhythmic foundation.
BASE Layer:- The rhythm section supports the complete 'mix'.
The FILLER: Between the layers is a 'glue' that stops the layers sliding off and holding the one above in the right place. This 'glue' can be likened to relationship between the audio elements and the relationship between the people involved.
The ICING: Effects - reverberation and delay that can pretty-up and give 'space' to a mix. Without any 'icing' the mix can still be tastey. Too much 'icing' and the mix becomes sickly and takes away from what is at the 'core'.

MUSICIANS SHOULD PRESET THEIR INSTRUMENT PATCH LEVELS IN SUCH A WAY THAT THE AUDIO ENGINEER DOES NOT NEED TO ADJUST INSTRUMENT CHANNEL GAINS DURING A SERVICE

"Anyone can make sound. Not everybody can make music."
- St Vincent

Pan / Balance

1. Onstage sources can be seperated in a mix using pan/balance so the centre image is not so cluttered.
2. Pans that are too wide can mean audience on one side of the room may not hear audio panned to the other.
3. Pan can be used to balance the acoustic image on stage e.g. an electric guitar amp on left panned slightly right.
4. Stereo signals e.g. drum overheads or stereo keys are often panned hard left and right. If stereo overheads are not panned hard left and right, what is in the middle (the snare) may appear louder when the signals are combined, as the snare signal is common to both microphone signals.



Mixing Foldback

Musicians and singers should be within the horizontal and vertical flare of the speaker horn to hear higher frequency content.

Setting Up Mixes for Foldback Wedges and In-ear Monitors

- Keep the stage and wedge volume as low as possible while allowing musicians/singers to hear what they need.
- Vocalists often pitch off the bass.
- Instead of increasing the volume of one instrument in the foldback, it may prove better to lower the volume of the other instruments.
- The monitor mix will normally sound different to the front of house mix. Foldback operators should not be expected to change foldback levels during a song.
- With their polar patterns in mind, place microphones in such a way as to best reject sound from the nearest foldback wedges. Avoid returning condenser signals from choir or headset microphones back through foldback wedges.
- Potential feedback frequencies can be reduced using an EQ.
- Hearing damage: Check in-ear and musician volume on stage (particularly the drummer). In a caring manner we should encourage our musicians and singers to preserve their long-term hearing. *See the companion Hearing Management Handbook.*
- In-ear monitors allow for audio communication from an MD. Initial users of in-ears may feel disconnected from the congregation. An auditorium mic mixed in may help. In-ears can be wireless.Wired versions are less expensive. A digital console may be required to provide the most efficient ways to provide multiple in-ear mixes.

Volume Tools

Changing the volume on a mixing desk channel can be achieved **manually** by raising and lowering the **volume fader or gain knob** (or as a result of EQ changes). Volume changes can also be **automated** through **dynamic processors.**

on-stage volume
\+ acoustics
\+ amplified mix
\+ congregational sound
= OVERALL VOLUME

For the operator to mix effectively, the front-of-house sound level must be **above the acoustic level**. To find the acoustic level, drop the front of house level during the rehearsal. Note how the foldback's 'sound' from the audience position.

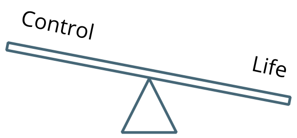
If the unamplified mix is too loud - then the onstage volume and/or room acoustics need to be addressed.

Dynamic Processors

Dynamics - the difference in volume between the quietest and the loudest sounds.

When above or below a chosen volume **threshold, dynamic processors** automate volume functions. These dynamic processors are **inserted** onto a channel or group and affect the audio on that channel or group. Dynamic processors include the **compressor, gate, limiter, de-esser** and **downward expander**.

The simplest dynamic processor is manual finger-on-volume fader:
Threshold: At what volume you lift or lower the fader.
Ratio: By how much you lift or lower the volume.
Attack: How quickly you lift or lower the volume.
Release: How quickly you return the volume to the original level.



Automating volume using dynamic processors can help to control difficult tasks. Overuse of dynamic processors can be fatiguing to hearing and reduce the amount of **'expression'** and **'life'** the final mix contains.

COMPRESSOR - Reduces the dynamic range between loud and quiet sounds making the volume changes more 'even'.

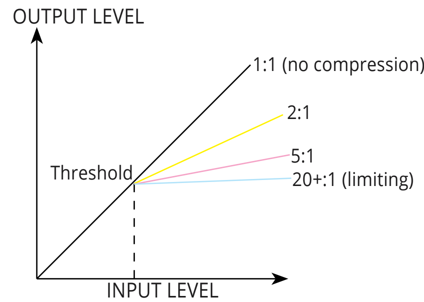
When to use:

- If a singer is too loud at times and too quiet at other times, a compressor can provide a more consistent volume, that also helps make the vocalist easier to 'blend' with other sources.
- When passing around an MC microphone between different people.
- To improve the volume consistency of a vocalist with poor mic technique, or on instruments that require a more consistent volume: e.g. kick or snare, or acoustic guitar,
- To catch quieter audio sources without being startled or discomforted by other sudden louder sounds.
- As an 'effect' to help provide 'glue' to tighten the band or elements within the band, making volumes 'smoother'.

The most used compressor parameters:
Threshold: Above a nominated volume level the compressor will reduce volume.
Ratio: How much the audio above the threshold will be reduced.
Gain/Output: Increases the channel volume to compensate for any compressor reduction.

The reduction LEDs give an indication of how much the processor is reducing the sound level.

How to use a compressor:



- A voice may need just subtle dynamic control at a ratio of 2:1. Wind the threshold back until there is some reduction shown in the LEDs. Balance the amount of automatic volume control required with how much expression and 'life' to allow. Try not to compress more than 9dB.
- For an MC mic that gets shared quickly, an operator may choose to heavily compress the mic signal 'for a short time' to maintain as consistent a volume as possible at a ratio of 3:1 or more, at times allowing over 9dB reduction. Use the faders to also control the volume. When the mic is no longer shared, return the ratio to a 2:1 more subtle compression.
- For a snare or a kick drum use a ratio of 3:1 or 4:1. If the instrument is very dynamic and more automatic control is required, wind the threshold down till the volume is more even.
- A wildly dynamic bass player who 'slaps' and 'plucks', may require a higher compressor ratio of 5:1.

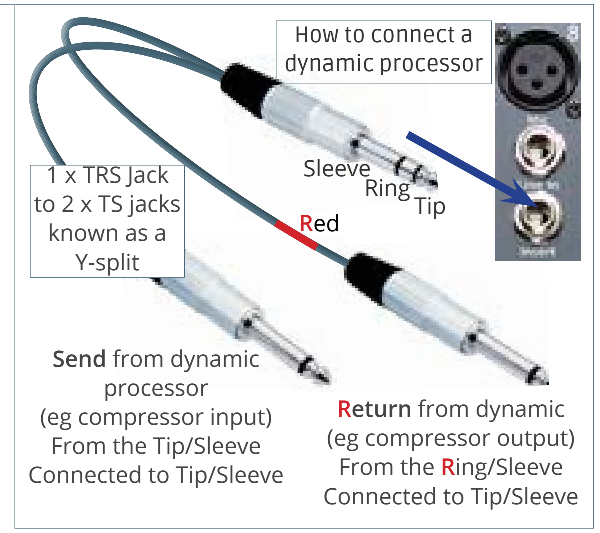
LIMITER - An extreme compressor (ratio 20:1)

To provide safety to equipment. In a 'mix' a peak limiter can be used to increase the overall volume by heavily cutting audio peaks, allowing the gain to be increased without 'clips' showing on the audio meters. Overuse can sound like distortion.

GATE - Reduces the volume when the audio level is below a threshold.

Useful for: reducing drum resonance on a kick, snare or tom from 'ringing on' too long. A gate can also be used to remove a troublesome buzz heard on an instrument when the volume of the instrument is not playing. Careful setting is required to make the gate close and open in a natural way so as to not be distracting or being shut more than is desired.

Gate parameters:
Threshold: Below this volume threshold the gate will reduce or shut off the volume.
Ratio: The ratio of how much the audio below the threshold will be reduced.
(Other parameters include **hold** and **release**)



Effects

- The use of reverb can give a 'dry' sound a sense of space.
- The use of tap delay in time with a song, can give a sense of depth with a rhythmic element.
- Excessive use of effects become a distraction.
- Putting a reverb and/or delay on a singer who is out of tune makes bad notes last longer.
- Critical listening to commercial recordings provides a reference for appropriate use of effects.

Reverberation and delay effects are a mix of orginal sound + effected sound

EFFECTS

original sound
- flat sounding and close

EFFECTS

original sound + delayed copy
by delay volume same as original
- loses clarity

EFFECTS

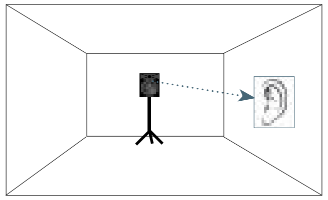
original sound + delayed quieter copy
by delay volume down lower than original
- has depth clearly heard original and delay

EFFECTS

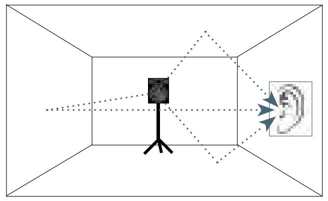
original sound + delayed quieter blurrier copyby delay volume down lower than original and high cut EQ
- has depth clearly heard original

EFFECTS

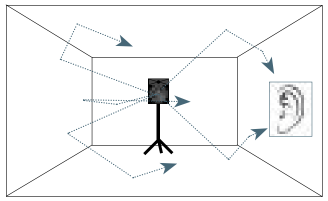
original sound + 4 delayed copies
by delay with too much 'feedback' at same volume as the original
- loses clarity



Direct sound



Pre-delay: the first early reflections from bouncing off walls, floors and ceiling etc



Other reflections after **reflecting** off two or more surfaces

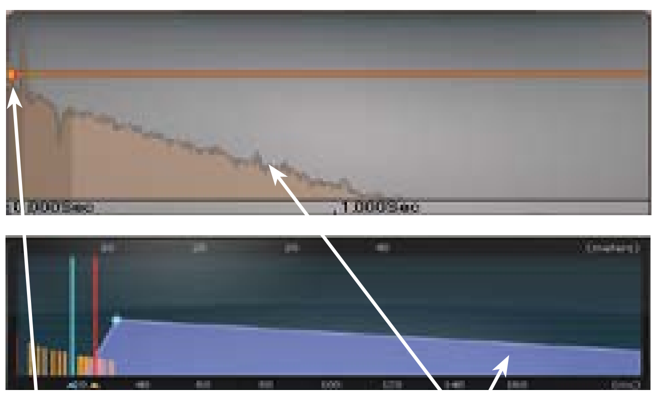
Reverberation Parameters

Time is the length of the reverb tail (how long the reverb will be heard). Keep the time short enough to not detrimentally muffle subsequent sounds.

Predelay. Using a predelay of 20-30ms delays the reverb slightly allowing the effect not to affect the clarity of the original sound.

Diffusion creates a smoother spread of reverb. Less diffusion contains more noticeable delays (reflections).

High-Frequency Damping affects the sound in a similar way as the effect of hanging up drapes in a room.



Early reflections Tail/Rev Time

Commonly one auxillary feeds into the effects unit and outputs just left or stereo left and right

How to connect:

1. On a channel, **SEND** some of the sound **TO** an effects unit using a **postfader auxillary OUT** to the **Effects IN**.
2. Return the signal from the effects unit **OUT** to the console via a normal channel or **auxillary return**. Make sure the returned channel doesn't also send that signal back to the effect, or feedback will occur.

The most commonly used parameters:
Reverb: Reverb type eg hall, plate, room; Reveb time/tail e.g. 1.7 seconds.
Delay: **Time** e.g. 1 sec (best to use a tap button to match the tempo with the delay). **Feedback** - selects how many repeats of the sound.

Analogue vs Digital Mixing Consoles

Analogue Mixing Consoles:

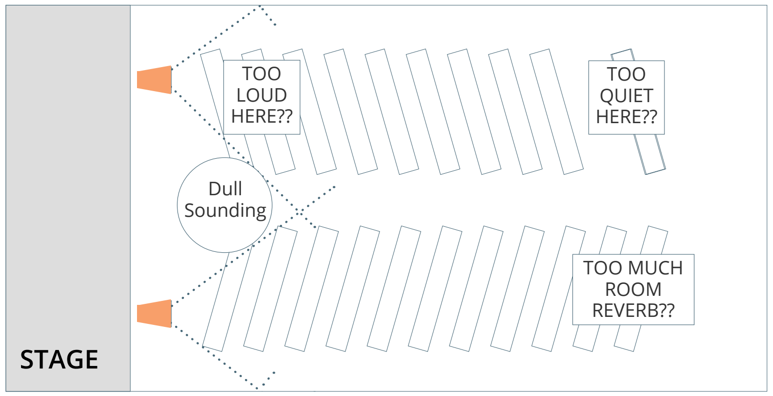
- Perceived ease of use,
- Less expensive for a basic mixing console,
- All functions are visible, and
- Dedicated controls for each channel.

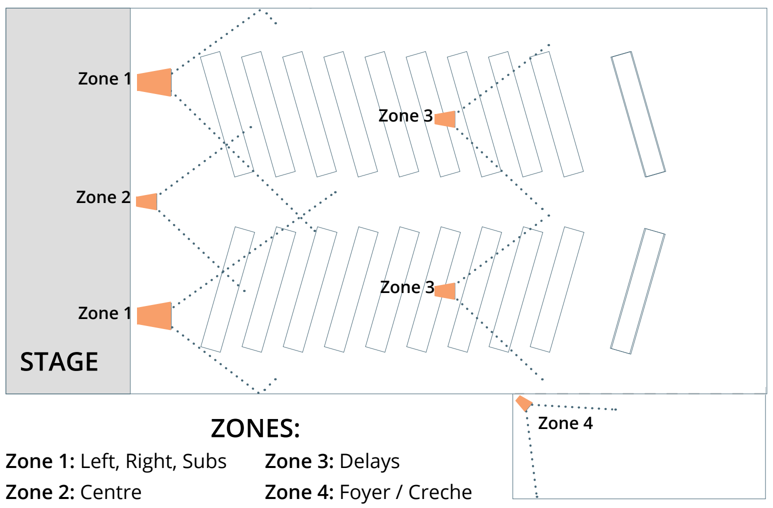
Digital Mixing Consoles:

- Smaller footprint,
- Presets can help with EQ and dynamics,
- Multiple use controls accessible through a channel select button,
- Built in effects and dynamics,
- More options for in-ear monitoring,
- More grouping options,
- Scene save/recall options for quick setup per user,
- More routing options,
- Multitrack and stereo recording/ playback options,
- Digital snake options,
- Wireless control,
- Multiple layers may cause some confusion.

Acoustics and Speaker Placement

Too loud in one area while too loud in another? Hard to hear things clearly because of room reflections? The sound is dull sounding in some places within the room?





ZONES:

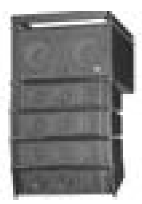
Zone 1: Left, Right, Subs
Zone 2: Centre
Zone 3: Delays
Zone 4: Foyer / Creche

Extra speakers can allow greater consistency of sound within a room without 'blasting' some people and being too quiet for others. These speakers can create **'zone's'** of coverage and may include: left/right FoH, centre fill, front fill, side fills, delay speakers, creche and foyer mixes.

Different levels of music, vocals or presenter mic can be set differently for each zone, and can be adjusted using a zone processor or using matrixes on a mixing console.

Careful setup can help to focus attention psychoacoustically towards the centre of the stage.

Alternatively a line array system may provide a solution for even coverage without the use of delay speakers



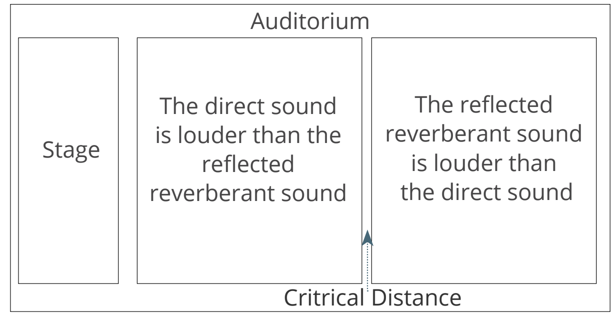
Considered placement of each speaker is important especially when using multiple speaker systems. The initial system set up requires an experienced technician to independantly adjust sound for each 'zone' by manipulating volume, tone, time alignment and phase through an audio processor.

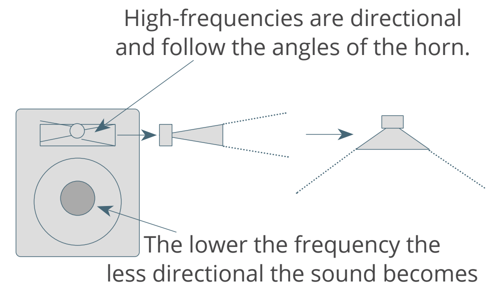
An example of Zone Controller (dbx ZonePro 1260)

Natural reflections in a venue create reverberation that can either be beneficial or detrimental depending on the building use:

- Classical instruments prefer favourable large acoustic reverberation obtained from buildings with harder surfaces like wood, or concrete. These venu types may conversely make speech less intelligible.
- Speech and contemporary instruments require less reflective surfaces that reduce any natural reverberation. These treatments include carpets, heavy drapes, soft chairs and specialised acoustic treatments.

Different parts of the building may exhibit different levels of reverberation.





Adding more speakers at a lower volume level can provide greater audio consistency in a room without 'exciting' the natural room acoustics too much.

Hearing Aid Loops:

Since 2004, NZ law requires systems to assist the hearing impaired for any new buildings with gatherings of 250 people or more. This law also applies in any theatre, cinema or public hall or assembly spaces, and in old people's homes occupied by more than 20 people.

Induction loops are one of three assistive listening devices designed for occasions whenever the hearing aid is not sufficient. Radio (FM) or infrared (IR) systems require the temporary provision of non-personalized receivers that need to be stored, checked, charged, cleaned, repaired and replaced at some point. Induction loops work directly with the individual's personal hearing aid using its built-in receiver (T-coil) to transmit the sound.

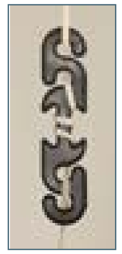
Induction loops can be laid in piping within the concrete foundation, on the surface of the concrete or in the ceiling. Setting a cable into the concrete means cable cannot be cut accidentally by carpet installers or from the movement of chairs etc.

A mixing console feeds the induction wire with a signal to a specialized amplifier. The induction wire has a 3m coverage but for maximum benefit and for minimal extra cost, a two loop overlapping system running at 1.5m spacings apart with a perimeter 1.5m from the walls is an efficient setup. Often a cancellation loop wire is run near the stage to avoid interference from stage equipment.

Hearing Loop
Switch hearing aid to T-coil







Further Training

Hearing Management and Live Sound:
www.classaudio.co.nz

Audio Essentials for Church Sound by Chris Huff
- eBook and online resources with a church focus:
www.behindthemixer.com

ProSoundWeb: Website with articles relating to church sound
www.prosoundweb.com/church

Mixing With Your Mind - Michael Paul Stavrou

Master Handbook of Acoustics - by F. Alton Everest & Ken C. Pohlmann

CMSS
COMPTON MICROPHONE SUSPENSION SYSTEM

Other Useful Software Resources

QLab from Figure 53: Simple and reliable audio playback - brilliant!! Free for stereo version. figure53.com/qlab/

WavesTracksLive from Waves:
Free and easy to use multitrack recorder and multitrack playback!!
http://www.waves.com/mixers-racks/tracks-live#presenting-tracks-live

Audacity: Free and Open Source recording software
http://www.audacityteam.org

Smaart 8 Di from Rational Acoustics: Paid software for analysing the frequency content of sound
http://www.rationalacoustics.com/smaart-v7-di/

Presentation software for churches:
Easyworship: https://www.easyworship.com
ProPresenter: https://renewedvision.com

Apps for portable devices:
Spectrum analysis and audio tools: **AudioTools**
SPL/Dosemeter: **NIOSH SLM**
Scripts and programmes: **PDF Expert, PDF Editor**
For musician music words and chords: **OnSong**
Musician Tools: **Cleartune, RODE Rec, Pro Metronome**
Guitarists: **Guitarist Ref, Amplitube, Guitar Toolkit**

Collaboration, Experience and Creativity