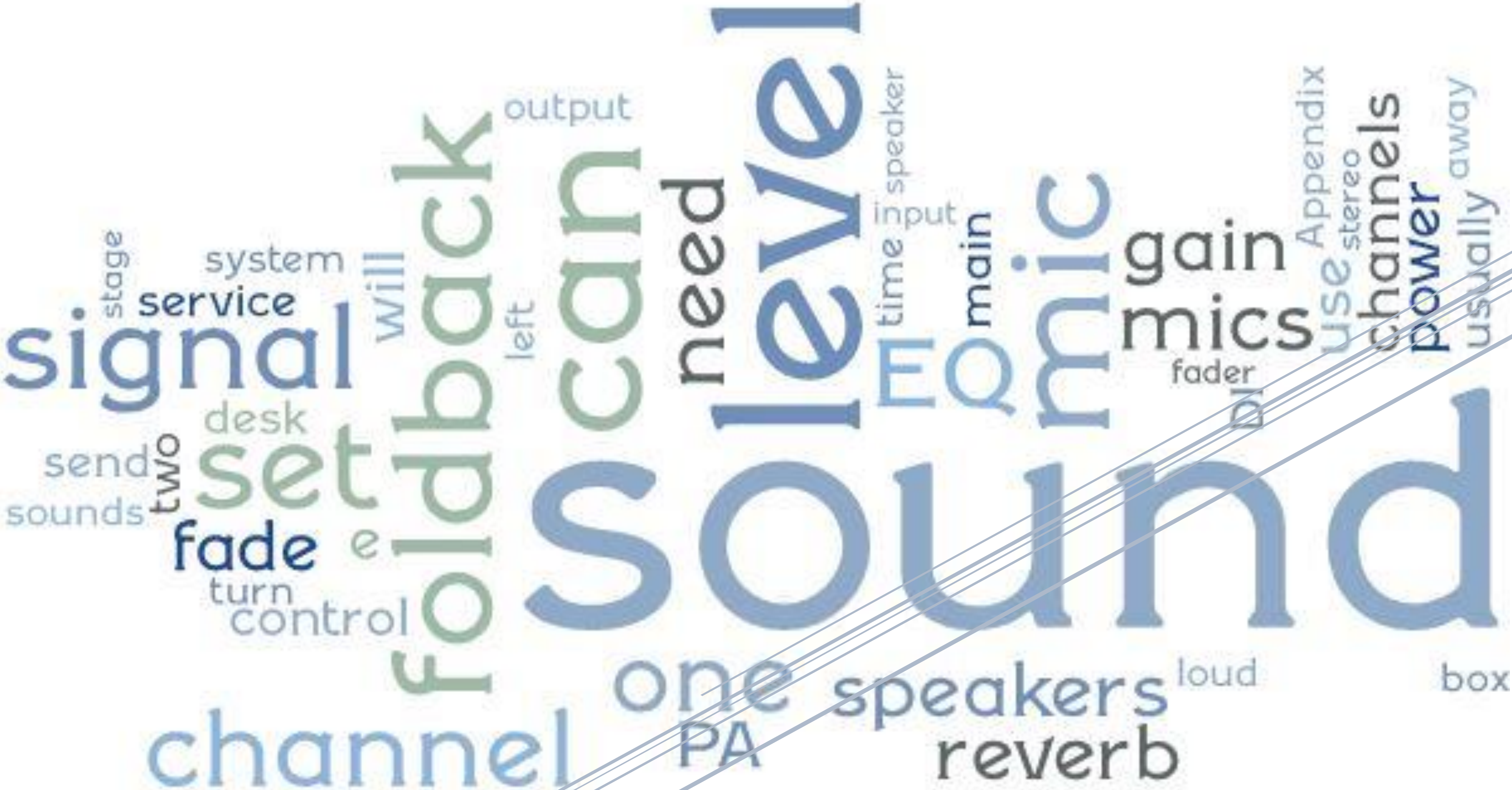
CGBC SOUND MANUAL

September 2015



Croxley Green Baptist Church
www.croxleybaptist.co.uk

Croxley Green BAPTIST CHURCH

Devoted to Christ, His People, His Mission

Sound Manual

This sound manual is written with the intention that a new person could find out all they need to know to operate the PA system in the main church sanctuary.

Over the years church worship has been enhanced by technological developments and new equipment. However there comes a point in the life of any church when that equipment and technology becomes so complex that potential volunteers are put off by the steep learning curve involved. It is also the case that different operators may use the equipment in very different ways (as there is usually more than one way to do things) which can be confusing. This manual sets out a common approach.

In a large room with lots of bodies absorbing sound, some amplification is essential. This is what the PA system is for – to **reinforce** the sound so that it can be clearly heard by all. Some of our congregation are elderly and others have hearing difficulties; if the service is too quiet, they cannot fully participate. On the other hand, tinnitus sufferers may find their condition aggravated by excessively loud, unpleasant sounds; we want to avoid both extremes so that the services can be enjoyed by all.

The aim of this manual to make operating the PA system as simple and transparent as possible, to achieve the best results we can so that the technology never gets in the way of the worship.

This manual is formatted in 'landscape' A4 to allow easy reading on a computer screen as a PDF. **Alan Kerry**

Contents

The ‘essentials’ are in the main text, the (optional) appendices go into greater detail, which may appear daunting – do not fear, it really is simpler than it appears!

INTRODUCTION

"Do nothing out of selfish ambition or vain conceit, but in humility consider others better than yourselves. Each of you should look not only to your own interests, but also to the interests of others." *Philippians 2: 3- 4 NIV*

Musicians can be idealistic, vague and insecure individuals. Engineers like precision, predictability and control. As a 'musical engineer' I am only too aware of the tensions this can generate, and hope that this manual will go some way to encouraging constructive ways of working that minimise stress!

If we start from the premise that musicians are creative, 'ideas' people, and engineers have the technical know-how to present this clearly to a wider audience, we may be better able to value each other's gifts and abilities.

With this in mind it is important to acknowledge that any attempt at a definitive manual necessarily represents one viewpoint. This is both a strength and a weakness. It is a strength in that some consistency of approach is helpful, but a weakness if it smacks of arrogance and isn't open to newer, better suggestions.

I therefore invite ALL feedback (except the audio kind!) on how this manual can be improved, or to point out the no-doubt numerous errors it contains. The content has been gleaned from years of reading Sound-on-Sound magazine, from two very helpful tutorials from Joel McGuinness (sound engineer at New Wine) and from plenty of practice mixing sounds in the luxurious safety of a studio setting. Studio work is a doddle compared to live sound, because if you get it wrong you just do it again, live sound has no such safety netting. I do NOT pretend to be an experienced live sound engineer myself, so there are almost certainly errors and omissions.

Thanks for taking the time to read this.

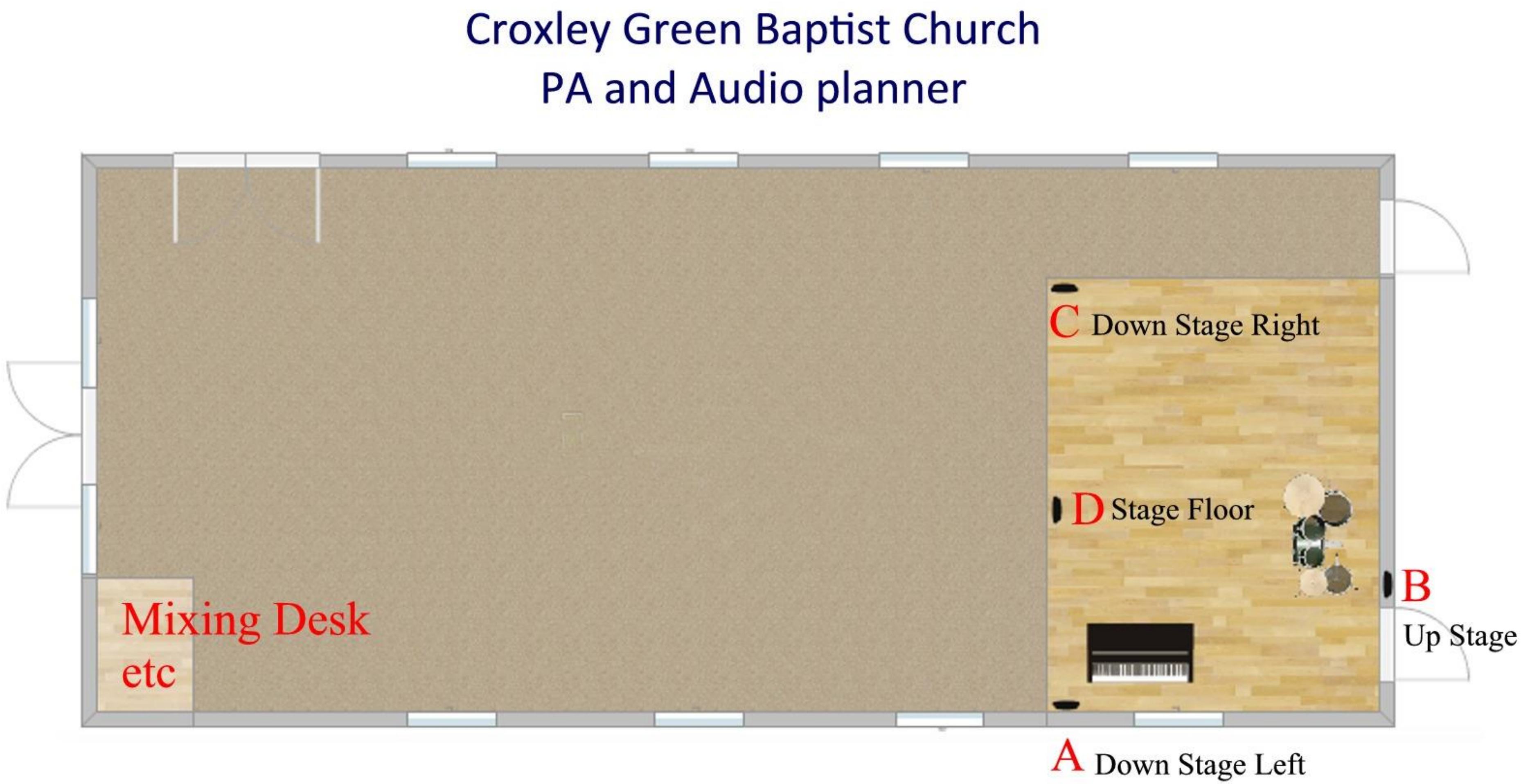
Glossary

PA	‘Public Address’ system, the whole setup, mics, mixer, amplifier and speakers.
Signal	The sound that you are using the PA to reinforce
Channel	The route that a signal follows through the mixer, allowing processing along the way
Processing	Changing the signal in some way (gain, EQ, effects)
Gain	The amount of amplification added to the signal to make it louder
EQ	Equalisation – ‘tone controls’ for adjusting the level of certain frequencies within a sound (‘bass’, ‘middle’ and ‘treble’ for instance)
Reverb	Reverberation; an artificial ‘echo’ effect added to a sound (usually a singer) to make it sound nicer!
DI Box	Direct Injection box, used for plugging instruments into the PA at microphone level
Phantom Power	Low level DC voltage needed by some microphones or DI boxes if they contain active circuitry
Bus	A destination for an audio signal, to allow it to be combined with others (‘mixed’)
Pan	A control to ‘shift’ the signal from one bus to another, e.g. from left to right.
Aux	An auxillary bus for creating ‘auxilllary’ (i.e. additional or different) mixes to the main output mix
Send	Routes a channel to an auxillary bus for ‘sending’ it out of the mixer.
Return	Brings an external signal back into the mixer (e.g. after external effects have been added).

Audio Connections

Inputs

The church building has a large number of potential audio inputs and outputs to allow flexibility and to reduce the number of long wires trailing all over the floor.



The connections at the following locations are usually:

A = DSL (Down stage left)

1 = Brown microphone

2 = Blue mic

3 = Red mic

4 = Green Mic

5 = (Spare input)

6 = DI box from Piano

7 = (Spare input)

8 = Output for foldback 1

B = US (Upstage)

1 = Guitar 1 (usually taken from DI box to allow DI box to send 'link' to a guitar amp on stage)

2 = Guitar 2 (as above – maybe bass guitar?)

3,4,5,6,7 = spare

8 = Output for foldback 2

C = DSR (Down stage right)

1 & 2 = Inputs from Audiobox – USB sound box for plugging in a laptop at the front for laptop sound out.

3 – 8 = (Spare inputs)

D = SF (Stage Floor box)

The Lectern mic and optional Baptistry mics are connected here and appear at the desk as SF1 and SF2

The Mixing Desk

The desk is a Soundtraks Topaz Maxi 24 which has 20 mono channels and 2 stereo channels. When the system was installed the leads were labelled with the conventions of ‘upstage’, ‘downstage’, ‘stage left’ and ‘stage right’ **as viewed from the ‘performer’**, hence they say DSL1 or US7. I have stuck with this convention and relabelled some of the wires that were missing tags, but also labelled the stage boxes A-D for simplicity as above.

Outputs

The Mixer has 4 group buses 1 to 4, to which individual channels can be mixed, the group mixes should then be routed to the MAIN L&R bus outputs. It is best **not** to route the channels directly to L&R but via the group buses instead. The four group outputs feed:

1. Front speaker (‘left’ if thinking in stereo – and using pan control)
2. Rear speaker (‘right’ for pan)
3. Welcome area ceiling speakers
4. Line out to rear hall (requires an amplifier in the hall – or a powered speaker).

There are a number of controls for each vertical channel strip – essentially the signal goes in at the top, and is then ‘processed’ in order until reaching the fader at the bottom – see the channel strip overview on the next page:

TOPAZ MIXER – Channel Strip

XLR mic input
Line input
Insert socket **(do not use)**
Direct channel out socket **(do not use)**

+48V Phantom Power switch (for condenser mics or DI boxes only)
Pre-amp Gain Control - the most important knob!
HPF - (high-pass filter switch) removes all low frequencies below ~ 80Hz
EQ Section comprising 6 knobs, black is for cut and boost, white is to select the frequency:

- HF (high-frequency) – cut or boost up to 15dB at 12kHz
- MF1 (high-mids) – white to select between 350hz and 15kHz, black to cut or boost.
- MF2 (low-mids) - white to select between 35hz and 1kHz, black to cut or boost.
- LF (low-frequency) – cut or boost at 80Hz

EQ Switch - when this is OUT the EQ section is bypassed
Auxiliary 1 send (adjust PRE-FADE level sent to Aux 1)
Auxiliary 2 send (also PRE-FADE)
Auxiliary 3 send (can be switched PRE or POST fade)
Auxiliary 4 send (can be switched PRE or POST fade)
Auxiliary 5,6 (or 7,8 - always POST-FADE level)
Switch to toggle the Aux 5,6 knob to send to Aux 7,8 instead)
Pan Control – think 'front' and 'rear' rather than 'left' and 'right'
Pre-Fade Listen switch - sends the pre-fade signal to the headphones and the LED level indicator lights
Mute switch - mutes the channel output, (but NOT the PRE-FADE level Aux sends – i.e. foldback)
Peak indicator light - lights if the input signal is too loud (clipping)
Switches to assign to any or all of:
Main L&R output (yellow fader)
Group 1&2 output (red faders)
Group 3&4 output (red faders)
Fader (grey) - optimal sensitivity is at 0dB, three quarters of the way 'up', but you can add 10dB boost above this.

Scribble strip - to label individual channels for easy identification

Channel IN	Source	Notes
1	Brown mic, DSL1 (down stage left) = A1	Some of the vocal mics have their own on-off switches, check these are 'on' if there is no level. These 'dynamic' mics do **not** require phantom power.
2	Blue mic, DSL2 = A2	
3	Red mic, DSL3 = A3	
4	Green mic, DSL4= A4	
5	DSL5 = A5	
6	DSL6 = A6 = Piano DI box	Needs phantom power
7	SF1 = Lecturn mic (condenser mic)	Needs phantom power
8	Lapel radio mic 1	Check batteries every time
9	Lapel radio mic 2	Check batteries every time
10	Handheld radio mic 1	Check batteries every time
11	Handheld radio mic 2	Check batteries every time
12	Guitar 1, US1 (upstage)= B1	If this is taken from a DI box then phantom power is required
13	Guitar 2, US2 = B2	
14	Front Laptop SF1 = C1	
15	Front Laptop SF2 = C2	
16	SF2 = 'Baptistry'	Condenser mic is best for this (Phantom power)
17	Spare	
18	Spare	
19	Spare	
20	Reverb Return (from the Yamaha reverb unit in the rack)	To adjust overall reverb level
21/22	Stereo channel – for computer output or MP3 player etc	
23/24	CD Player	
FX 1 LR	Unused – designed for effect returns but no EQ control	Could be used as additional stereo input in required
FX 2 LR	"	"
FX 3 LR	"	"
FX 4 LR	"	"
Tape in	Phono connectors	But no EQ control

Output	Feeds	Notes
Group 1	No output connected	– internally feeds to LEFT MASTER bus
Group 2	No output connected	– internally feeds to RIGHT MASTER bus
Group 3	SUPP – Feeds ceiling speakers in Foyer	Driven by Australian Monitor AMIS120P amp
Group 4	Line out to Hall	For feeding sound to the large rear hall.
LEFT	Master Out – feeds to Front speakers via Ultragraph	Driven by QSC USA400 amplifier
RIGHT	Master Out – feeds to Rear speakers via Ultragraph	"
MONO	No output	
Tape Out	Phono connectors, connect to external recorder	'Mirrors' the master out mix
AUX 1	Foldback 1 (front), DLR7 = A7	Pre-fade send level (see below)
AUX 2	Foldback 2 (rear), US8 = B8	Pre-fade send level
AUX 3	Foldback 3 if needed, US7 = B7	Pre or post-fade level (switchable)
AUX 4	Hearing Loop amplifier	Switch to post-fade (see below)
AUX 5	Unused	
AUX 6	Unused	
AUX 7	Reverb send	Always post-fade level
AUX 8	Unused	

AUDIO-VISUAL QUICK CHECKLIST

SUMMARY:

1. ***TURN UP***
2. ***OPEN AV desk – POWER ON (AUTOMATED SEQUENCE, turn the key only),***
3. ***DETERMINE WHAT IS REQUIRED FOR THIS PARTICULAR SERVICE (Mics, stands, backline amplification, video? laptop? other sound sources).***
4. ***SELECT ACTIVE CHANNELS and KILL UNECESSARY CHANNELS***
5. ***PULL UP MASTER FADERS (Red and Yellow to 0dB)***
6. ***ADJUST GAIN FOR EACH CHANNEL (red rotary knob)***
7. ***ADJUST EQ for CLARITY AND WARMTH***
8. ***SET UP RECORDING (See Appendix A)***
9. ***SET UP VIDEO OR OTHER SOUND SOURCES – CHECK THEY WORK, sound and vision.***
10. ***KEEP ACTIVE DURING THE SERVICE* - Tweak, Cycle, Ride, Walk around**
11. ***PACK AWAY AT END OF SERVICE, POWER DOWN (and foldback speakers)***

For each service you should aim to:

1. ***TURN UP*** – nice and early (9:00am for 9:15am service, 10:30am for 11:00am service)
2. ***OPEN AV CUPBOARD (keys in photocopying office – number 4) – POWER ON –*** The [Furman PS-PRO II] Power conditioner / sequencer takes care of powering everything up in sequence, while avoiding power surges. Therefore all other switches should be left ON all the time, and the Furman used to switch everything else on and off with the key. Separately turn on computer, projector and power switches on foldback speakers at the front of the church.
3. ***DETERMINE WHAT IS REQUIRED FOR THIS PARTICULAR SERVICE*** (Mics, stands, instruments via DI boxes, backline amplification, video, laptop etc). Radio mics are stored in the drawer – check battery level on the transmitter by powering on. There are two lapel mics and two handheld radio mics. The lapel mics can be swapped for headset mics.
4. ***SELECT ACTIVE CHANNELS and KILL UNNECESSARY CHANNELS.*** Decide which channels will be used. Phantom power is required for channels which come from a DI box (guitars, piano) or condenser microphones (e.g. the lectern microphone) ***KILL REDUNDANT CHANNELS*** by turning down the GAIN, this reduces the risk of feedback and muddy sound if the fader is moved accidentally. In addition you can mute these channels but note that MUTE only mutes the main send, it does NOT mute the foldback which is 'pre-fade send'. Killing the gain is preferred for this reason.
5. ***PULL UP MASTER FADERS*** – set these to 0db level for left, right, master, then adjust individual channels as below.
6. ***ADJUST GAIN FOR EACH CHANNEL*** – Again, set the fader to 0dB – usually this is where you want it to be, then it is the rotary GAIN knob that you adjust when setting up the level for each channel. This way, all the levels are roughly equal, then during the service you can tweak the relative levels with the faders, which will mostly be around the 0dB range where they are most sensitive. If there is no signal then work systematically from the source to the desk, is the mic switch on, is it properly connected? Which wall or floor socket is it plugged into? Is that line plugged into the correct desk channel, does it need phantom power (DI box or condenser microphone), is the gain up? Is the channel routed to the correct output bus?
7. ***ADJUST EQ for CLARITY AND WARMTH*** Cutting problem frequencies sounds better than boosting frequencies. The feedback range is often around 500Hz, boxiness around 700Hz, harshness 1-2kHz,). The singer and pianist will probably be very close to their mics but the speaker may be a good couple of feet away from theirs (especially bible readers!). The

‘proximity effect’ means that the bass response of most microphones drops off exponentially as you move further away, so the EQ and gain settings might be very different for singers and speakers simply because of how far from the mic they stand. Aim to use EQ controls to achieve CLARITY and WARMTH of overall sound, but try to avoid excessive boost. In terms of frequency, speech is essentially consonants and vowels (‘please Carol’), the vowel sound or chestiness of a voice around 300 Hz is very important for intelligibility, as is the ‘ssh tck pss’ of consonants in the 3-6kHz region. These frequency ranges are just a guide, but a common error is to have too little bass in the voice which makes it just as hard to understand as too little treble. However, only Barry White creates anything below 100Hz so leave the hi-pass filter switch in place for anything other than bass guitar, kick drum or keyboards. See frequency chart Appendix A for more on this.

8. ***SET UP RECORDING– see Appendix A***
9. ***SET UP VIDEO AND OTHER SOUND SOURCES –*** if there is a video played from the front laptop then the Audibox USB device is used. This can occasionally be problematic if the laptop has not loaded the drivers properly. Rebooting the laptop with the box plugged in usually fixes this.
10. ***KEEP ACTIVE DURING THE SERVICE –*** Tweak the EQ for speech to get a warm but crisp sound, cycle through the individual sounds in your head during songs. Ask yourself, can I hear that particular instrument or singer? Is it too loud? ‘Pro’ live sound engineers are usually busy people, especially at the beginning, live sound rarely reaches a point where you set up the levels and leave them alone. Ride the faders, i.e. pull down vocal faders if they’re not singing, this reduces ‘spill’ and helps to clean up the sound. Walk around – if at all possible, move away from the desk during a song and get the sound ‘first hand’ as the congregation are hearing it from the centre of the room. Another important tip is to avoid the temptation to mix according to the headphones, these are useful for troubleshooting (why can’t I hear the guitar?) but should NOT be used to shape the sound. What sounds good on ‘phones might sound terrible through the PA and vice versa.
11. ***PACK AWAY IN REVERSE ORDER AT END OF SERVICE and POWER DOWN*** – When powering down the PA, simply use the key on the Furman unit to switch everything off in sequence and to avoid power surges. In addition don’t forget to switch off foldback speakers, projector and computer.

APPENDIX A - How to record a service on Zoom H2N

This is setup to record to TWO stereo tracks, one is from the stereo mics on the unit itself (records congregational singing etc) and the other comes from the line in source, fed by the Mixing desk ‘tape out’ (which is essentially the same as the main mix output).

Plug in the power (mini-USB) and minijack leads

The minijack comes from TAPE OUT on the mixing desk

Power on – slide switch on side for a second then release

Check 4Ch recording is activated with the rotary selector on the top

Press Rec – the counter at the top starts counting straight away

(The recording level can be adjusted here, but usually leave at about ‘4’)

Don’t worry about where the recording goes in terms of folder and file name – it all ends up on the memory card and is copied via USB to edit on computer and create an MP3 or burn a CD. Also don’t worry about starting the recording too early, or switching off – the memory can store a few hours quite comfortably, which is very easy to edit down.

Inputs		
Radio mics	Two lapel mics (can be switched to headset mics) with clip on belt transmitter units Two hand-held mics, with internal transmitters.	
Lectern mic	Condenser microphone – more sensitive and does not exhibit the 'proximity effect'	Needs phantom power. Rather more delicate, do not drop.
Singer Vocal mics	Shure SM58s and the like. 'Proximity effect' means bass frequencies are louder when the source is closer to the mic.	Tough as old boots!
DI boxes ('direct injection') 	Useful for a wide range of inputs, e.g. guitar, keyboard, submixer. The DI boxes need phantom power from the mixer (or a 9V battery but phantom power is better!). They match the impedance of the input to a mic level so that a line level signal can be sent down a much longer lead than would otherwise be possible without signal degradation. The input is via a Jack socket, the link socket allows you to 'split' your input, i.e. one split (the output on the XLR socket) goes to the mixing desk and the other (the link jack socket) can connect to the guitarist's own amplifier. The pad switch attenuates (turns down) the signal by 20 or 40dB – usually for high level keyboard outputs, Ground lift helps to eliminate hum if there's a ground loop.	

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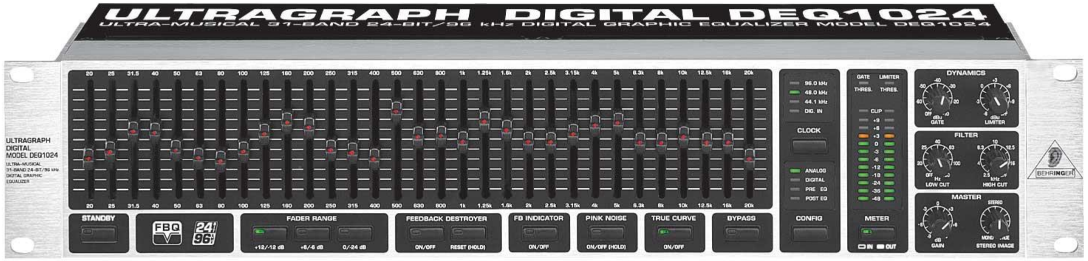
<tr><td>Outboard rack</td><td>

Behringer Ultragraph DEQ1024
Graphic EQ and Feedback suppressor, and dynamics processor
The main function of this is to ‘tune’ the PA system to the room using the 31-band Graphic EQ to tame or boost frequencies where the room response is not flat. Ideally this is set up using pink-noise and a reference mic. It should be set up ONCE and left alone.


However there are a couple of other controls to be aware of.
The dynamics section includes a GATE control. Essentially this mutes the whole system unless the signal is above a threshold, determined by this control. If the gate is set too high then there is a possibility that very quiet music might not get through, e.g slow fade in on a video sometimes.

There is also a control for ‘stereo width’ which could cause confusion – (see section on ‘Mono or Stereo’), this should be set to fully wide for our system to allow separate control of front and rear speaker levels.</td></tr>
<tr><td>Reverb Unit</td><td>Yamaha Rev 100 – Fed from Aux7, returns to channel 20. No need to adjust, provides a simple plate reverb, nice on vocals, the mix balance knob should be set to fully ‘wet’ so you adjust the amount of reverb with Channel 20 fader.

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'Insert' and 'Direct'

Channels 1 to 20 accept input from mic (XLR) or line (jack). There are two other jack sockets for each channel and it is important to understand that we **don't** use these. An 'insert' jack is a way of plugging an external effects unit in series with the input signal, so it goes from the preamp to the external unit and then back to the channel strip for EQ and level (fader). It is a stereo jack socket that connects to both an 'in' and an 'out' for that channel, so if you plug an external sound source in you WILL get some output, but it bypasses the gain control so you have less control over the level. Also if you plug a mono jack into the 'stereo' insert socket you might interrupt the signal chain and effectively 'mute' that channel. The 'direct' jack is designed for multitrack recording. It provides an output 'directly' from the pre-amp, i.e. before EQ, pan and fader, so it gives the 'cleanest' signal for recording purposes. It will not work as an input.

A note about foldback

Musicians need to hear themselves, and this is provided by foldback speakers. We usually use just two. Of course these create sound that is also heard by the congregation. The PA person can turn down the level coming out of the main speakers but should leave the foldback signal reasonably loud so the musicians and singers can hear themselves. This is sometimes an area of conflict – musicians should be encouraged to play at a comfortable level, but not TOO loud, and sound men need to appreciate that they do not have 'total control' over the sound coming off stage.

Mono or Stereo

The PA speakers are hung centrally at the front and back of the church – it is obvious that this is not a 'stereo' system, but in fact many live sound venues that have speakers on either side of the stage are in fact run as 'mono' systems anyway. In other words the left and right signals are identical, and 'panning' from one side to another doesn't do anything. In our system Group 1 feeds the front speaker and Group 2 feeds the rear speaker so pan controls, or group 1 & 2 faders can be used to adjust the balance

between front and rear.   Note that this only works if the Behringer output is set to 'wide'; for a while it was (wrongly) turned to 'mono'.
When plugging an external 'stereo' source in, use Channel 21/22.   If you need to use a mono channel then just use one jack, don't plug the second one into the 'insert' or 'direct' sockets.
Stereo channels have 'balance' controls instead of 'pan' – shifting the balance fully to one side only increases the level of that side, any signal in the opposite channel is lost.

Pre-fade vs Post-fade level sends

Basically a pre-fade send is unaffected by the fader, whereas a post-fade send is affected.   There are two common situations that highlight the correct use of these:

1. Stage Foldback speakers – the musician / singer still needs to hear themselves even if the sound man thinks it would be better to spare the congregation!   So foldback sends should always be 'pre-fade' (Aux 1&2)

2. Reverb – adding reverb to a vocal mic should be proportional to the level of that channel, so a 'post-fade' send is required.   Also the hearing loop level balance should mirror the 'main out' levels, so is 'post-fade'.

The PFL switch routes the Pre-fade level for that channel to the headphones and LED monitoring section so you can assess the INPUT (pre-fader) level of a single channel at a time.

In addition the master Auxillary sends are accompanied by AFL (after fade listen) switches, so you can monitor exactly what is being sent to those outputs in the same way.

Appendix D - A Guide to Successful Live Sound (adapted from Joel McGuinness lectures at New Wine)

Setting gain:

The most important aspect of mixing is setting the gain correctly, if the signal is too quiet at this stage it'll never cut through, and if it's distorting, you can't salvage it later on. So you need to take some time setting the gain while you're getting a representative signal from the input (singer, guitarist etc.) For example, you can't set it with a vocalist whispering 'one-two, one-two' a couple of times, you need them to be singing as they will in the service. Really you need to set the gain at the loudest point of the practice. These loud peaks should just about light the red PEAK LED, most of the time the signal should be a healthy green level on the LED indicators. Use PFL switch to send the signal to the main LED level indicators and it should be well up the greens, sometimes lighting the yellows. If it's only lighting one LED then turn up the red gain knob at the top of the channel in question (NOT the fader at this point).

You need to set the level for EACH channel at EVERY service. If a channel isn't being used it's a good idea to turn the gain down to zero. This reduces the chance of **feedback loops** through unused channels, as well as encouraging the next engineer to set the gain level themselves when that channel IS needed.

The desk cannot be setup and left from one week to the next (though one frequently encounters church PA systems with masking tape saying 'Here' and 'This Level' and 'DO NOT TOUCH'!!).

Once the gain is set, leave it alone for the rest of the service, getting the sounds balanced is the job of the faders, and will vary from song to song, speaker to speaker. If the gain is set correctly then the faders give a visual representation of how loud each channel is in relation to each other.

EQ - some tips

HF	Very rarely cut Maybe boost **gently** to add clarity
High mid	Maybe cut around 6kHz to reduce sibilance ('sSSSssS') and harshness Boost gently around 3kHz to add presence to vocals (brings them 'forward')
Low Mid	Cut around 500Hz to reduce boxiness and thin the sound, or to tame 'booming' feedback Rarely boost – feedback zone, but excessive cut removes warmth.
LF	Cut to reduce boom and thump (unless it's supposed to be boomy and thumpy, like, ooh I don't know, kick drum perhaps!) Gently boost to add fullness if tinny (assuming those frequencies exist! Female vocals don't go much below 200Hz)

Stage Setup

You can't turn a poor performance into good sound, and as the engineer, it is sometimes your job to suggest when the music isn't working. Some suggestions you might make are:

- Band layout – rhythm section together (bass and drums need to hear and see each other to play together)
- Encourage eye contact between vocalists and leader
- Mic positioning – to avoid feedback, keep vocal mics close to mouth (in my experience pretty much ALL vocalists need reminding to get closer to the mic, in live sound you can pretty much never be too close to the mic). 'Mic technique' is where the singer backs away a few inches when they sing loudly and leans in closer for quieter passages (a kind of manual level control or compression). It takes practice, it's not a natural thing to learn, and a more common problem is too much movement to and from the mic. Sound levels drop off exponentially with distance so if you're usually 1 inch away and move to 2 inches away it will

halve the volume, 4 inches away is one eighth as loud, by the time you've stepped back from the mic the difference could easily be 20-30 times quieter. Keep the rear of the microphone facing the monitors, and don't hold around the basket (affects the 'directionality' of the mic).

- Foldback monitors should be set back a bit (we don't have ears in our feet!) pointing at your head, without music stands in the way (Joel McGuinness calls the type of metal music stands we use 'nuclear reflector shields!')

Feedback

Is never good! Turning down the gain and turning up the fader achieves nothing, except spoiling your visual reference for how loud anything is. There are only three solutions: Move the mic (away from reflective surfaces), reposition the speakers, or use EQ (or fourthly – does it really need to be that loud? PA is REINFORCEMENT of speech, we are not trying to make the sermon audible in The Artichoke public house!).

Foldback

The Joel McGuinness method of setting foldback levels is very quick and easy, and should give a good mix for the two foldback speakers we currently use. If there's not enough or too much, the WHOLE foldback MIX can be turned up or down on the foldback speaker itself, rather than fiddling with individual foldback levels at the desk.

Basically you always set the auxillary send at one of three positions only:

1 Fully left (i.e. OFF for channels that don't need any foldback- this is very important if you remember that the foldback is PRE-FADE, so otherwise unused mics can still generate foldback, which creates muddy sound at best and feedback at worst)

2 12 O'clock – 'background' level (e.g. set lead vocal level on the guitarists foldback to this level, and vice versa)

3 3 O'clock – 'My mix' level (e.g. the level of lead vocal foldback on the Vocal foldback auxillary)

Let the band try it out with the main PA off (i.e. master faders down). This is the direct sound that the band is making, and it's probably quite loud, especially with no bodies in the church to absorb the sound. The faders are then used to REINFORCE this sound, and it is quite possible that you won't need to put everything through the main speakers to reinforce this basic sound.

It is NOT possible to set the sound for the main PA first and then add foldback. In other words, the band will set it's own level which you will NOT be able to turn down on the desk. If it really is TOO loud, then tell the band and check that the foldback is giving enough signal. Musicians tend to PLAY louder if they can't hear clearly enough (which masks everyone else, who then also play louder and so the level spirals up.) Getting a comfortable foldback mix is the key to getting the musicians to relax and play clearly and confidently, without struggling to play louder. In other words, paradoxically, if the band is playing too loudly, turning UP the foldback may help (and letting them know!).

So what needs reinforcing i.e. which faders need to go up? Certainly vocals, and probably guitar. The kick drum (if mic'd) can really help to bottom out the sound, a balanced sound (one that covers the full frequency range) sounds much more natural and pleasant than an unbalanced sound of all mids and highs. Psychoacoustically, an unbalanced sound will sound harsh and LOUD and unpleasant at a much lower true volume (SPL) than a balanced sound, so try to find some bass from somewhere, kick drum, keys maybe?

Try to use all the elements to construct a balanced overall sound, some bass (below 150Hz), some warmth (300Hz region), some tune (700-2000Hz) some clarity (3KHz) some definition (6kHz ish) and some air (12kHz). Reading that will probably make you want to turn everything up but remember, EQ CUT always sounds more natural, and is less likely to create feedback. These are the frequencies you want the instrument or vocalist to provide, cut the others. For example, in one service you might want a violin providing the interest in the tune region, one vocalist might be mostly providing warmth as backing vocals. Ultimately you can't learn EQ by numbers. If we were all playing sine waves you could, but in reality every musical signal contains harmonic frequencies. EQ is therefore about correcting obvious problems and shaping a balanced overall sound, and the settings will vary enormously depending on context, so one band line up will require widely different settings to another. Practice, practice practice, and in the words of Joe Meek – if it sounds right it IS right.

Mixing – Top tips

Mixing is like playing an instrument

It's a full time job

- Keep scanning through the sounds in your head
- Always ride the vocal faders
- Be aware of solo instruments – bring them up to make them shine, then back them off again after the solo
- Snare drum and lead vocals should usually be the loudest elements in the mix
- Avoid overcrowding of frequency and space
- If you don't need it throw it away (cut unwanted frequencies, kill unused channels)
- Turn off effects when people speak!

REVERB

The church is a nice reverberant space anyway, why would we want to add any more? There are occasions where a sound can benefit from being a bit less ‘dry’, especially on foldback. Most singers sing better with ‘comfort reverb’ adding a short tail to the note. While singing a note, most of the sound comes from inside the singers head, but a reverb tail means they get to hear what they’ve just sung coming out of the system. This can be a huge help with a singer who’s struggling to stay in tune, it’s not just ‘more me’ they need in the foldback, it may be some reverb.

As an ‘effect’ to add to the sound, reverb is generally overused, but one or two elements (a violin or a flute) might be lifted by a good wash of reverb, providing the rest of the sound is solid and supportive. If everything is drowning in reverb it’ll all get lost. Reverb tends to push sounds ‘back’ in the sound field, making them sound further away. Also bear in mind that we rehearse in a (mostly) empty reverberant space, and in the service there may be up to 200 bodies absorbing that sound and those reflections, so adding a little reverb to singers can help.

The Yamaha REV100 reverb unit can do a lot more than basic reverb, but it’s probably best to set it to a simple plate reverb and add to selected channels according to taste, I think the best presets for vocal reverb are 8 or 12.

APPENDIX E – A rough guide to the sound frequency spectrum – EQ