AW4416 Audio Workstation Signal Flow Tutorial

This tutorial will help you learn the various parts of a CHANNEL by following the signal through INPUT #1. Use the Signal Flow Diagram included with this document. It can be found in the Operation Guide on page 24, inside the back cover of the Reference Guide or on the inside back cover of the AW4416 product brochure. Learning to follow the signal flow is a major step in mastering the unit. This article assumes you are looking at the Signal Flow diagram and are near the AW4416 so that you can locate the associated screens.

YAMAHA **AW4416 Audio Workstation**

Signal Flow Tutorial

Follow the Signal Flow block diagram - follow the signal through input #1. Channel 1 and 2 are the same. Channels 3-8 are identical with the exception of channel 8 which has a high-Z input.

Input jack to preamp – Our signal enters the physical input jack, which on input #1, can be either balanced TRS ¼" or XLR. On the block diagram a circle with 3 dots in it represents an XLR connector. The TRS ¼" jack is represented by 3 open dots, tip-ring-sleeve, (sleeve connected to ground while tip-ring connect on to the preamp (triangle). You will also see that this channel has a 48volt phantom power supply for condenser mikes.

Preamp and PEAK light – A triangle – always represents an amplifier – in this case the preamp responsible for matching incoming gain. The circle with the arrow through it (below it) signifies it is an adjustable amp. This is the Gain/Trim control. Next to it you see the PEAK light. It flashes when signal comes within 6dB of clipping. It is okay to see it flash on and off – you want to avoid it being on continuously.

> The Gain/Trim control is used to 'gain stage' or match the signal to the input. You want to match the level that the board is expecting to the output level of the device. If you think of the signal flow as water through pipes: then if you have a device that is –40 (like a microphone) you would need to adjust the fitting for the pipe. If on the other hand you have a +4 signal, you will need to adjust the fitting for the pipe. If you fit a small pipe (-40) to one set for a large pipe (+4), or vice versa you will have a problem. Either not enough signal will flow or you will overload. Noise will fill up the difference. There is SIGNAL or there is NOISE. Noise being unwanted signal. In the pipe analogy, there is water or there is air. You want mostly water in your piping system. This is why you want to see the peak light flash during signal flow, it alerts you that you are sending a good amount of signal but are not overloading. When you overload the peak indicator light will remain steady, OUCH.

INSERT I/O - Following this is the opportunity to insert an external product. On physical inputs 1 and 2 you have the INSERT I/O (input/output) jack. This insert point (a TRS jack) can be used to access an external piece of gear or to receive the output of an external microphone preamp, for example. This single jack can act as both send and receive. Use a typical "Y" Insert Cable (a single male TRS ¼" to two male ¼" mono – send/return) when you want to patch in an external device. Or it is ideal when connecting an external preamp - just plug it in here with a single ¼" cable (it bypasses the input channel's Gain/Trim preamp).

A/D to INPUT CHANNEL SELECT - Next is the 24-bit Analog to Digital converter. Up until now we have been working with the *physical*. From this point on the signal is in the *digital* domain. Next you will see the INPUT SELECT block – this is where you connect a physical input jack (#1 in our example) with a mixer channel, 1-24. Any physical input can be routed to any of the 24 channels. This means that you never have to re-plug a cable, simply re-route the signal via the Mixer Channel Input Assign.

> SCREEN: This is accomplished on the [**SETUP**], [**F1**] PATCH IN page: Mixer Channel Input Assign. If you want input jack #1 to show up on channel 10 you would put "AD 1" in the 10 place. If you want to keep it on the default channel, channel 1, make sure that "AD 1" is in the 1 place.

INSERT RETURN POINTS and INPUT METER – On the Input Select block you will see that signal can be routed through a channel from any of 46 different places. This includes the 24 possible inputs (optional boards have 8 inputs each), the 16 HDR (Hard Disk Recorder) tracks, the two stereo returns from EFF1/EFF2 and the stereo digital inputs. During the recording process you can listen to the signal as it is sent **to** the HDR. During playback of data you will select a channel to monitor the signal coming back **from** the HDR. / Next you see a METER.

> SCREEN: This Meter can be viewed on any screen that shows INPUT. This is either on a per channel basis or as a part of the total view of all inputs. For example, from [**HOME**], view MIXING LAYER [**1-16**]. If you select F1 [**1-24/RTN**] you can see the meter array for all 24 possible inputs or if you press the [**CH VIEW**] for Input 1 (press [**SEL**] on channel 1) you can view the first 2 channels. (This view will always show an odd-even pair. Pairing accommodates stereo inputs by linking 2 channels).

ATTENUATOR/PHASE/EQ – The attenuator allows you a way to adjust the signal in line, when necessary. This could be when combining signals within a channel or when you are bringing in external signals and you need to downwardly adjust the overall signal flow. Normally, this is left open at 0dB. / Phase reverse means you can accommodate inputs (typically microphones) that are wired differently. / This is followed by the 4-band, parametric EQ and another meter view point. Because you add and/or subtract dB with an EQ, another meter is provided so that you can continue to check the overall level at this point. You will find a switch that will let you change the meter view Pre or Post EQ.

> SCREEN: You will find specific views for Phase, Equalization and Attenuation. If you think of input signal as water going through a series of pipes, then an attenuator is like a valve on a pipe. Typically the valve is left fully opened to allow free flow of water. You only close it to reduce flow. The 4-band parametric EQ lets you fine-tune the tone of the signal. You can adjust the Q, F and G. Q is width (how many frequencies will be affected) A narrow Q, will have a high number (10), and will make a steep curve and effect a specific band of frequencies. A wide Q (0.1) will effect an large band of frequencies. "F" is Frequency. You can select the center frequency of effectiveness. G is gain, plus or minus 18dB. The EQ is handled with 44-bits to accommodate some pretty radical equalization. If you are EQ'ing pretty radically, then use your meters to ensure that you are not exceeding the bandwidth of the system. You have dedicated buttons and knobs for EQ. Simply [**SEL**]_Select the channel and access the EQ. You can have the screen automatically switch to the EQ or you can view the values, in miniature on the top line of the screen. (See UTILITY: Preferences).

EQ to Dynamics Processor – After the EQ and meter is the configurable Dynamics Processor. Each channel has a Dynamics Processor for controlling the level of signal automatically. A dynamics processor is a specialized amplifier to either decrease or increase the dynamic range of input. Dynamic Range is how soft to how loud a signal gets. This amplifier takes action when a particular signal level is reached. This level is called a threshold. When the signal reaches the threshold the amp does its function. Because the dynamic processors effect overall level (there is an output gain control) there is another meter that will show both gain reduction and the level post this stage.

> The Dynamics Processor (DP) can be configured as a compressor/limiter, expander/gate, compander, or ducker. When using these devices you **can** effect the resulting impact of the signal. However, when working with them recognize that the gain control is there to return the signal **level** to what we know is optimum level. It is not subjective. If you are using a compressor, for example, it will reduce the gain when signal exceeds the threshold. This will tend to reduce the overall level, depending on your threshold and amount of compression (Ratio) applied, the signal may now be less than optimum but the 'punch' is just what you need. You have the Gain control in the DP to restore the optimum level. Use the DYNAMIC ON/OFF switch to A/B (compare) your settings.

INPUT DELAY – This allows you to delay the channel by a specific amount of time (in tenths of a millisecond). If you are working with some digital device this can be a great tool allowing you to delay when the channel delivers its audio from .1ms (approx. 4 samples @ 44.1kHz) to 59.0ms (2600 samples).

> You can set the amount of delay per channel. Interestingly enough, you can use this to delay one channel of an unpaired stereo signal, a few tenths of a millisecond, to get an very 'enhanced' stereo effect.

FADER – There is a METER both before and after the main channel fader. You will find a switch that will let you change the meter view (Pre or Post Fader). Again you can see just what you have done to the signal at each step. (The fader is a rectangle with an adjustable arrow through it – listed as just LEVEL).

> The fader can represent different levels depending on the FADER VIEW that is selected. You can select one the 8 AUX buses or the HOME view - INPUT/RTN or MONITOR. The fader is an in line attenuator. Like a valve on a pipe. Typically, this valve is set full open (+0dB) during record. And is set 'to your taste' during playback.

PRE and POST FADER sends – Take a good close look at the lines that head perpendicular to the flow both before (pre) and after (post) the Fader. You can feed signal Pre- or Post-Fader to AUX buses. You will see a line that also can connect the signal to the Direct Outs for channels 1-16, or to the SOLO bus (Pre-Fader Listen or After Pan Listen).

PAN – Technically called the panorama potentiometer, the pan pot lets you send signal to the Left or Right Stereo Channel. During the recording process (INPUT) you can use the PAN to bus signal to the ODD or EVEN track of a paired input. During playback (MONITOR) you can place the signal in the stereo panorama to your taste.

> It actually does this by adding resistance to the channel you are panning away from. When you are ready to experiment with PAN look up in the Reference Guide about how you Gang and Inverse Gang paired channels – this allows wild movement of signal in the stereo field.

PRE and POST PAN sends – In a similar fashion to the pre and post fader sends you have the option of sending signal individually for each, L and R channels.

SOLO – The Pre-Fader Listen (PFL) and After Pan Listen selection will let you hear the SOLO function either centered (pre) or "in place" within the stereo field (post) – your choice.

> The Solo function **preferences** are determined in [**SETUP**]→[**SHIFT**]+[**F1**] to [**CHANGE TAB**] then select [**F3**] Solo Setup. Solos can be either RECORDING (does not interrupt signal being sent to the target) or MIXDOWN (what you hear is what is sent to the target). SOLO can be either "in place" (post the pan) or centered. You can also select to SOLO a single channel via the ON button or multiple channels simultaneously

AUX – Signal pre or post the pan and/or fader can be sent to any of the 8 AUX buses. Each AUX bus has a fader (rectangle with arrow labeled AUX).

> When you flip the faders to the AUX 1 view, you can then choose either AUX fader view [**1-16**] or [**17-24/RTN**] when working with INPUT signal (pre the HDR). If you press [**MONI**] you can view the signal sends for the monitor mix (post the HDR). Typically, signal is recorded 'dry' and it is the monitor mix that is sent to the effect processors. In mixdown, the effects are added to the monitor mix that becomes the final stereo mix.

BUS – If you look closely you can see that the signal can be optionally connected to any of the 8 buses (vertical lines). You can assign them using the [**ROUTING**] function. This gives you the ability to use the buses for re-routing signal like when bouncing tracks. When "Bouncing" you combine signals of various channels on any bus or group of buses and then route those buses to other record tracks.

Notice that to the right of the vertical lines for the BUS lines, Stereo L/R, Solo L/R, and AUX 1…8 you will see that the STEREO bus has its own attenuator, 4-band EQ Dynamics Processing, insertion points, meters and level controls. This is very important for Mastering, etc.

> You can view the bus level my pressing [**MONI**] and then [**F3**] Bus. This is very important when you are using a bus to combine signal from more than one channel. You have a 'software' (screen) fader to attenuate the signal for each BUS and/or AUX send.

Conclusion
We just followed the signal from a typical input through the channel. Next take a look at the signal flow of a Monitor channel (from the HDR through the channel). Learning to read a Signal Flow Block Diagram will help you become more confident in your patching and routing with the AW4416. On close inspection of the block diagram you will find routing for the RECORDER channels, 'the sampling pads, the CD-RW's audio input, the Digital inputs and outputs, the assignable OMNI outputs, the optional I/O boards, even the headphone, metronome and the monitor outputs. Refer to pages 24-30 of the Operation Guide for more details on the signal flow diagram. Refer to it when you want to understand how signal gets from one place to another. It is like a bus map, pun intended. You get on at one point and you travel through switches and connections to get to another point. Transfers are free, as long as there is a connection point. Don't get lost or worried when the signal goes off the page, just realize that it can be terminated by you (like when a bus is assigned to a track of the HDR) or re-inserted into another channel.

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AW4416 Block Diagram