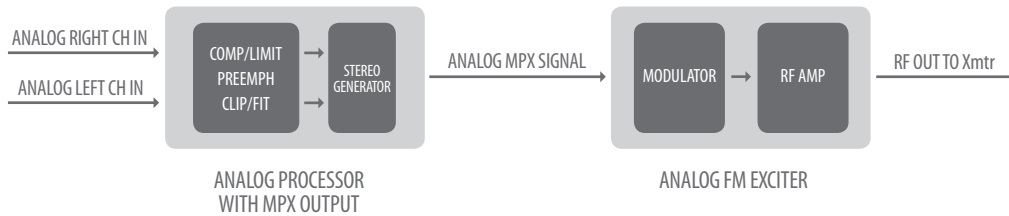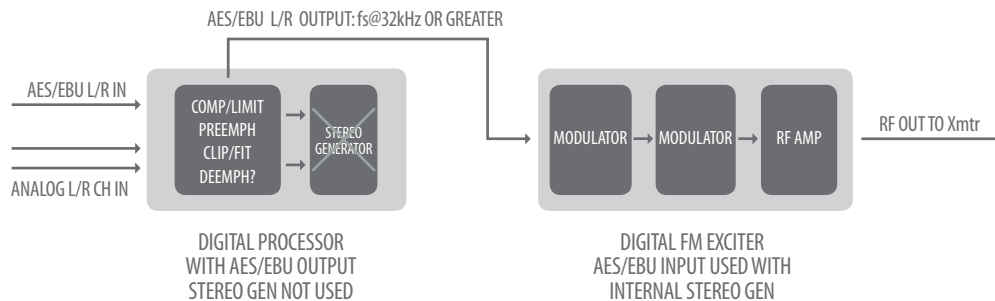# OMNIA DIRECT
MPX DIGITAL CONNECTIVITY

Prior to the advent of digital exciters for FM radio, integration of an audio processor (with integrated stereo generator) was done via a baseband or multiplexed (MPX) signal connected directly to the exciter's broadband (DC – 53kHz) input. The MPX signal would be routed directly to the modulator. This tried and true method is a present day standard. Any audio processor, be it analog or digital, that contains its own stereo generator offers an analog based MPX signal on a BNC connector. The connection of this signal, should it be directly out of the stereo generator itself or from the output of a composite STL system, is then connected to the MPX input on the exciter via another BNC connector. Anyone who has installed audio processing will understand this.

ANALOG RIGHT CH IN

ANALOG LEFT CH IN

COMP/LIMIT PREEMPH CLIP/FIT

STEREO GENERATOR

ANALOG MPX SIGNAL

MODULATOR → RF AMP

RF OUT TO Xmtr

ANALOG PROCESSOR WITH MPX OUTPUT

ANALOG FM EXCITER

With the introduction of digital exciters, the only MPX input is analog and it must be passed through an analog-to-digital converter (ADC) before it can be digitally modulated. The sampling rate of the ADC must be relatively high due to the wide bandwidth of the MPX. Consider for a moment the audio spectrum of FM is 99kHz wide. Therefore, any analog signal connected to a digital exciter must be sampled at 200kHz or higher. Once digitized, no other processing is required before modulation.
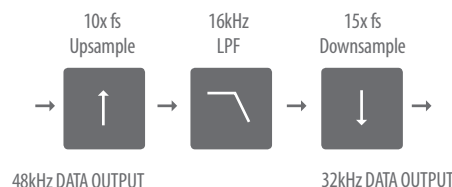
To date, if an all digital transmission path is desired, the only interconnect between the audio processor and exciter requires using the AES/EBU discrete left/right input. This path is quite a bit more complicated than the MPX path because it must condition the audio signal in preparation for stereo generation and herein lies potential for audio degradation. In this configuration, the FM-Stereo signal must be generated in the exciter.

AES/EBU L/R OUTPUT: fs@32kHz OR GREATER

AES/EBU L/R IN

COMP/LIMIT PREEMPH CLIP/FIT DEEMPH?

STEREO GENERATOR

ANALOG L/R CH IN

MODULATOR → MODULATOR → RF AMP

RF OUT TO Xmtr

DIGITAL PROCESSOR WITH AES/EBU OUTPUT STEREO GEN NOT USED

DIGITAL FM EXCITER AES/EBU INPUT USED WITH INTERNAL STEREO GEN

Experience has proven this method less than ample in modulation efficiency. This is primarily due to the required sample rate converters (SRC) employed in the exciter's input section. The signal that is arriving at the AES/EBU input of the exciter might be operating at a different sampling rate than the exciter is expecting. If so, a rate converter is employed to make the proper transition. This device can pose problems as the digital filter in the rate converter can generate overshoots to the already tight peak controlled audio signal that is being adjusted.

The following example illustrates how a synchronous SRC works. However, interfaces between equipment are not synchronous and therefore require asynchronous SRCs. These are functionally similar but considerably more complex. In order to synchronously change 48kHz to 32kHz sampling, the conversion is accomplished by scaling up, or interpolating the original sampling rate, usually by a factor of ten. Then, at the 10x rate of 480kHz, the signal is low pass filtered at the Nyquist of the new desired sampling rate. This filter is required to 'smooth' out the 10x rate. If filtering not used, aliasing products would result. Finally, the signal is scaled down, or decimated by the factor needed, in this case ÷15 to achieve the new rate of 32kHz. Below, is a block diagram of an SRC. While this sounds quite simple, and basically it is, there are a few issues to consider. Of main interest is the interpolation filter.

10x fs
Upsample

16kHz
LPF

15x fs
Downsample

48kHz DATA OUTPUT

32kHz DATA OUTPUT

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All audio processors, both analog and digital apply some form of overshoot control to the output filtering section. In most designs, this function is a form of integrated protection clipper working around the final low pass filter to obtain control.

In each case the overshoot component can be calculated as a product of what is known as the 'Gibbs Phenomenon', which states that an overshoot will occur at one-third the cut-off frequency of any low pass filter whenever a non-linear waveform is passed through it. In the case of broadcasting, the non-linear waveform would be that of a clipped waveform.

Knowing that the audio bandwidth used in FM-Stereo is 15kHz,

overshoot components will begin with any non-linear waveform above 5kHz. In this example, this would affect any signal above 5kHz that was clipped.
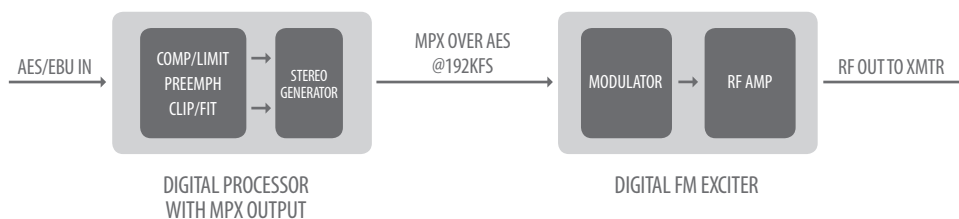
Should the slope of the previously described up-sampled interpolation filter appear greater than the slope of the final filter in the audio processor, then output overshoots may result in the sample rate conversion process! Unfortunately, these overshoots are generated after the processing unit. To remove them would require another limiting device. Some exciters provide an additional peak limiter, and the action of this limiter can degrade audio quality.

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The desired goal for the digital interconnect between audio processor and exciter, has been to emulate the older analog method, where the audio processor generates the FM-Stereo MPX signal, and then connect to a wide spectral input on the exciter. . .a digital BNC-to-BNC connection so to speak.

Omnia Direct generates the FM-Stereo MPX signal inside the

audio processor, and scales the sampling rate to 192kHz. Through recent firmware developments, the AES/EBU transom can accommodate a sampling rate at 192kHz, making it possible to transport the MPX signal directly from the audio processor to the input of the exciter. . .MPX over AES. Now it is possible to create a digital version of the older analog method. Below is an illustration of this concept.

AES/EBU IN → COMP/LIMIT PREEMPH CLIP/FIT → STEREO GENERATOR → MPX OVER AES @192KFS → MODULATOR → RF AMP → RF OUT TO XMTR

DIGITAL PROCESSOR WITH MPX OUTPUT

DIGITAL FM EXCITER

There are benefits to this configuration. Now it is possible to couple the output of the audio processor directly to the input of the exciter's modulator in a full linear fashion. There are no SRC functions in the path, which can degrade modulation and sonic performance due to overshoots. The absolute peak level at the output of the stereo generator, will translate to the absolute maximum deviation of the FM carrier. Accomplishing this insures the modulator will faithfully mirror the performance of the audio processor. This reduces, and basically eliminates any coloration of the audio due to the exciter's input and modulator sections.

Since the connection between the audio processor and modulator is as close to theoretically perfect as possible, there will be an improvement in sonic bass performance heard on the radio. This has always been a weakness within the analog MPX connection to an exciter, as any unwanted signal components near DC, would negatively affect the performance of the AFC (automatic frequency control) section of the exciter. This usually required the use of a high-pass filter, or DC-servo of some sort. Both of these can degrade the low frequency performance

of the modulator, which reduces or smears the aural texture of on-air bass.

Additionally, any specific functions unique to the audio processor/stereo generator are maintained, as they will not be forced to rely on any limitations imposed by the stereo generator in the exciter. By example, the alternative use of SSBSC (single sideband suppressed carrier) in the stereo generator, which has shown to reduce the effect of multipath, can be employed in the audio processor, and this alternative MPX signal will easily connect and operate correctly. Also, the AES/EBU signal is a standardized method. This now enables a digital MPX connection to be utilized as an industry standard.

Another benefit to this method is it now enables digital distribution of the MPX signal over wide broadcast networks. There are many such infrastructures of this type internationally. Any digital STL system capable of handling 192 kHz sampling should be able to carry this signal.

To date, Omnia Direct is available on the Omnia.11 and the Nautel NV series of FM transmitters.