

## 9.0 Using Digital Audio

### 9.1 Connecting AES/EBU Lines

The FM Pro digital audio interface operates with professional AES/EBU standard specifications. If your digital audio cables are somewhat lengthy, your cabling should use only 110 ohm twisted-pair shielded wire designed specifically for digital audio use. For very short runs of a few feet, standard twisted pair cable is usually acceptable. Keep cables as short and neat as possible to avoid noise pickup. Remember, digital audio signals are equivalent to radio frequency transmission and can suffer similar propagation anomalies. To avoid generating or receiving interference, pay attention to proper assembly of the XLR cable connectors.

#### 9.1.1 Cable Pinout

The following is the correct wiring pinout for AES/EBU cables: Pin 1 shield; Pin 2 positive signal; Pin 3 negative signal

### 9.2 AES/EBU Synchronization

The AES/EBU receiver will lock up to incoming sample rates between 25 and 55 kilosamples per second (KS/s). Once lockup occurs, the FM Pro's AES/EBU status screen will display the standard incoming rates of 32K, 44.1K, and 48KS/s. The received digital audio bitstream is subsequently fed to the analog-to-digital converter.

```

AES SIAIUS
=====
  Transmit rate: SLAVE
>Auto Source: OFF
  Rcv. rate: 44.1
  Rcv. error:
=====
Enter value.

```

Figure 9-1  
Processing, AES Status Menu

### 9.3 Digital-To-Analog Conversion

The FM pro utilizes a no compromise 20-bit digital-to-analog converter to receive the AES/EBU digital input signal and generate high quality analog for audio processing. We realize that, at the present time, there is virtually no chance that full 20-bit digital audio will be available to the FM Pro. However, developments presently in progress will soon open the door to vast improvements in broadcast digital audio. The FM pro will be

ready for all such improvements. Furthermore, use of a 20-bit a/d converter assures there will be no appreciable noise or distortion generated by the converter itself, leaving the question of audio quality entirely up to the digital audio source.

### 9.4 Auto Source

If Auto Source is turned on, the FM pro will automatically switch over from the digital to the analog input when digital audio data errors are detected. Error levels detected, in order of severity, are as follows :

- Validity bit high
- Confidence flag
- Slipped sample
- CRC error
- Parity error
- Bi-phase coding error
- No lock

You can set which error level is the lowest level that will trigger the Auto Source. The Auto Source will trigger after 10 error hits are detected within a 1 second interval. The Auto Source will therefore not switch over for the minor data errors which are normal in many digital STL's. After an Auto Source trigger, if no more errors are detected for an additional 4 seconds, the Auto Source switches back to accept the digital input.

If you are using the Auto Source mode, then you need to have a suitable analog input available to continue the broadcast program. Many users will maintain an analog STL or land line feed to the FM Pro as a backup to their digital audio link. If no analog input is supplied to the FM Pro, then when Auto Source switches to the analog input, your FM modulation will go silent.

**Note: To select the lowest error level you want to trigger the Auto Source, move the cursor pointer to Auto Source in the AES/EBU menu and turn the Spin dial to scan through the error list, stopping at the desired indication. To turn the Auto Source off, turn the Spin dial until "OFF" is indicated in the data error list.**

### 9.5 Digital Output

The FM Pro's digital audio output is generated by a no compromise, drift stabilized, 20-bit analog-to-digital converter. You can select asynchronous output sample rates of 32, 44.1, and 48KS/s, or

you may slave synchronize the digital output to the digital input.

Many users will see the benefit of asynchronous digital output. For example, if your digital STL runs at 32KS/s, 16-bits, you can come into the FM Pro at that rather inferior rate and resolution, process the sound in analog thereby elegantly reconstructing the audio waveform, and resample out at 48KS/s, 20-bits. Your digital FM exciter will happily accept the higher sample rate and resolution making your FM transmission all the better with much lower distortion and spurious content.

**Note: Thanks to an Apex patent pending design, the digital output of the FM Pro is free of all dc drift and bounce. All thermal offset drift of the a/d converter has been eliminated. Furthermore, the dc offsets of the analog audio stages are fully regulated by dc servo control. There will be no long or short term frequency drift or bounce of the FM exciter caused by the FM Pro.**

### 9.6 Facts About Digital Audio, Bit Rate Reduction and Dynamics Processing

Digital audio is a sampled and quantized approximation of the original analog sound. The higher the number of bits used to digitize an audio signal, the more closely it's reconstructed analog signal will resemble the original. The number of quantization levels available in the digital domain is a measure of the digital resolution of the audio signal. Too few quantization levels results in harsh distortion known as "grunge" or "splatter". There can never be too many quantization levels.

16-bit quantization has become a standard resolution in most of today's digital audio systems. Although this moderate digital precision has been found generally satisfactory for mass consumption, many people find that the presently available 18 and 20-bit digital audio is vastly superior in terms of realism and the reduction of noticeable digital audio anomalies. The reason is that, even if large amplitude waves can be quantized adequately in the 16-bit domain, small signals reflecting significant sonic details may be grossly underquantized or even lost.

The sample rate of digital audio directly affects the precision of the reconstructed sound as well. A

low sample rate, i.e., 32KS/s, results in a higher noise floor, a more spurious noise characteristic, and a narrower audio bandwidth. Higher sample rates improve not only those factors, but also permits better signal processing within the digital domain. It is desirable to have the highest sample rate possible and the largest number of bits possible.

The transmission and storage of digital audio requires a data bandwidth directly proportional to sample rate and geometrically proportional to the data size (number of bits). System economies forbid extravagant quantizing and sampling of digital audio, and compromises must be reached. Unfortunately, the compromises needed to make digital audio practical for most purposes are severe. For this reason, not only is the digital audio found in broadcasting usually of low sample rate and quantizing (32KS/s, 16-bits), but there may also be lossy data compression applied.

Lossy data compression, for example Musicam or Apt-X, will bring down the required data bandwidth, but causes further degradation of the sound. Whether the degradation is audible depends on the listener's criticality, of course, and upon many other factors. Multiple compression passes may be encountered in a complex distribution path and each compression stage may build upon the last causing severe damage to the sound. For this reason it is advisable to always avoid using compressed data pathways.

As stated before, the sound quality of digital audio improves as resolution is increased. It is therefore good practice to maximize the analog signal input amplitude to the analog-to-digital converter in order to obtain the maximum available digital resolution. However, the potential danger of driving too close to maximum input level is digital overload, a most unpleasant sound. To create sufficient headroom, standard practice is to establish the 0VU reference level of digital audio at 18 dB below peak clip. With each bit comprising 6dB of the available dynamic range below clipping, 0VU signals thereby receive only 13dB quantization. Small but important signals 20-30dB down from reference become only 8-bit audio. It is not hard to see why much digital audio can sound harsh and gritty. Furthermore, once a signal is poorly digitized there is no way to increase its resolution.

The quality of digital audio can be noticeably

improved by raising the 0VU reference level closer to clipping, and by using a high quality analog brick wall peak limiter such as the Aphex Dominator to prevent audio peaks from ever exceeding the a/d clip level. Of course, the Dominator could also be used to raise the density of the sound, getting a very loud and highly resolved digital conversion, if desired. When 18 or 20-bit digital audio is converted to 16 bits, the lowest bits are either truncated or dithered to noise. With either method, it would be best to maintain as much of the audio in the higher bits as possible to maximize resolution. Therefore, an 18 or 20 bit analog-to-digital conversion system also benefits from the Dominator.

Linear digital audio requires wide spectrum for transmission and large space for storage. For example, one channel of 16 bit sampled at 44.1kHz translates to more than 700,000 bits per second. The requirement for ever larger drives and wider transmission systems has been answered in part by various methods of bit rate reduction. Although sometimes called “data compression” they are all ‘lossy’ systems, which means that resolution is reduced, thus the noise floor and distortion are increased while subtle (sometimes not so subtle) details of the audio are lost.

One of the prime methods of all the systems to reduce data is the use of the psychoacoustic principle of masking. Essentially, the theory states that a higher level signal will mask lower level signals within a certain frequency range surrounding the higher level signal frequency, depending on the relative levels of the signals. The higher the high level signal is in comparison to the low level signals, the wider the masking frequency range. The frequency range of the masking effect is called the “critical band”. The bit rate reduction system discards the supposedly masked signals in the critical band and retains mainly the masking signals.

While there are differing opinions regarding the audibility of the artifacts of the various bit rate reduction schemes, the controlled listening tests upon which the proponents have obtained acceptable results have employed reproduction systems which were as flat as possible. There were no dynamics processors such as audio compression and equalization placed in the reproducing system. This means that data compression techniques are not designed with the anticipation that any dynam-

ics processing will be used. In broadcasting this can be a problem.

Dynamics processing, by definition, will change the level relationships between high level and low level signals. Since the most common form of broadcast audio processing is dynamic range compression, the high level signals will be reduced in level in comparison to the low level signals. Multiband audio processing will continuously change the masking relationships of the sound. This explains why data-compressed digital audio subjected to broadcast audio processing will often become exceptionally dirty and grungy because the masking algorithm presumed for data compression becomes greatly violated.

It would seem that if one must use a data compressed digital audio link, it would be better to place the link after the FM Pro, rather than before. In that way the FM pro could receive uncompressed digital audio, process it, then send the processed audio out to the digital link whereby the digital data compression would not be subjected to further audio processing. The problem with this proposal is that the highly processed audio from the FM Pro is too dense to fit the supposed masking parameters for inaudible digital data reduction. The data reduction processor has to make the data fit the available bandwidth one way or another, so, having no reasonable masking opportunities, it makes whatever severe and disastrous alterations to the digital audio that are necessary. It is, in fact, better to place the data compressed digital audio link ahead of the FM Pro. This arrangement, although still undesirable, usually yields better sound than data compressing the FM Pro’s output.

If a data compressed digital audio link absolutely must be used ahead of the FM Pro, then maximizing the quantization level by use of the Dominator to control audio peaks and a Compellor to ride average gain in the analog domain ahead of a/d conversion can mitigate some of the audio degradation of the data reduction system, and give the FM Pro a better quality digital audio input.

One final note about using data compressed digital audio. The digital algorithms of data reduction used in digital STL’s will sufficiently modify the audio input waveform to cause considerable peak overshoot. This will not be a problem when the digital link is placed ahead of the FM Pro since the FM Pro will re-limit the peaks prior to transmission.

However, if the STL is placed after the FM Pro, the STL digital audio output will not be suitable for direct connection to a digital FM Exciter. For these reasons and reasons already given, we recommend placing the FM Pro at the transmitter site in all cases involving a data reduced digital STL.

**Note: If a linear (uncompressed) digital STL is used, then it theoretically would be possible to maintain the FM pro at the studio and send the AES/EBU output up the STL directly to a digital FM exciter. At the time of this writing such an STL is available only on a T-1 wideband link. However, some T-1 equipment does rate conversion or other signal processing. You need to test your link for digital overshoot prior to making a commitment to this operating method.**

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