Z/IP ONE

The world's most advanced IP codec

Z/IP ONE is the newest addition to the Zephyr family

An affordable, 1 RU codec designed to help you get the best possible quality from public IP networks and mobile phone data services, even from connections behind NATs and firewalls. Agile Connection Technology (ACT), a Telos exclusive, is the foundation for the Z/IP's excellent performance on real-world networks. It delivers reliable audio despite varying network conditions, and without the need for user intervention. The Z/IP dynamically adapts to the network, minimizing the effects of packet loss and jitter.

When the network is well behaved...

You will benefit from the lowest possible delay and the highest possible fidelity. Should network conditions become challenging, the Z/IP ONE automatically responds by lowering the bitrate and increasing the buffer length, doing everything possible to ensure audio makes it to your studio reliably.

Z/IP ONE extracts excellent quality from even not-so-excellent IP connections

Thanks to a new codec based on low delay AAC: Advanced Audio Coding-Enhanced Low Delay (AAC-ELD), which gives excellent fidelity at low bitrates with nearly inaudible loss concealment and very little delay. And of course, Z/IP ONE speaks fluent Livewire; in addition to standard I/O, the Livewire connection lets it connect to any Axia IP-Audio network using just a CAT-5 cable.

FEATURES:

- Z/IP ONE is wireless capable and can connect to IP networks via Wi-Fi, EVDO, and UMTS.
- Exclusive Agile Connection Technology (ACT) automatically senses network conditions and adapts codec performance to provide the best possible audio.
- Largest choice of high-performance codecs: AAC-ELD, AAC-HE, AAC-LD, MPEG Layer 2, MPEG 4, AAC LC, MPEG 2 AAC LC, G.711, G.722 and linear PCM.
- Dual IP ports for separate streaming and control.
- Easy browser setup via built-in Web server.
- Push mode for one-way network connectivity such as satellite broadcasts.
- Multiple push mode, push to multiple destinations.
- Sophisticated NAT traversal support.
- Convenient directory server, no need to know another device's IP address.
- Transparent RS-232 channel for audio side channel or metadata, e.g., RDS.
- 8-bit parallel GPIO port for signaling and control.
- Slim 1 RU form factor fit is equally at home in a studio rack, remote kit or road case.

Input Levels Menu Selectable
+4dBu line level
-50dBu nominal microphone level

THD+N
< 0.03% @ +12dBu

Freq Response
+/- 1dBu 25-22kHz

Headroom
18dB

Dynamic Range
87dB unweighted
90dB "A" weighted

Crosstalk
>80dB

Output Clipping
+22dB

Calculated Output Impedance
50 Ohms, Differential

Calculated Input Impedance
6k Ohms, Differential

Analog to Digital Converter
24bits

Digital to Analog Converter
24bits

Conformance and Compatibility
Conforms to N/ACIP (Open) Standards.
Fully supports Session Initiation Protocol 2.0 (SIP).
Compatible with TCP, UDP, DNS, Zephyr Xstream, Uncompressed PCM and other Internet Protocols.

Codecs
SIP: G.711, G.722, MPEG Layer2, MPEG AAC, MPEG 4 AAC LC, MPEG 2 AAC LC, Linear PCM.
MPEG AAC-Enhanced Low Delay (ELD).
High Efficiency AAC.

Input Power
0.12A @ 120VAC 14.2 Watts