Z/IP ONE

THE IP CODEC THAT DROPS JAWS. NOT AUDIO.

OVERVIEW

Meet the world's most advanced IP Codec

These days, you can get broadband Internet just about everywhere, which makes it ideal for live remotes. But public Internet is also notoriously erratic. You could be lucky enough to get a good connection, but it might deteriorate during your broadcast. What to do? Cross your fingers and hope for the best? Or reduce your bit rate, sacrificing audio quality in hopes of making it through your show?

With Z/IP ONE, there's no luck needed. Z/IP ONE is a single-space rack unit IP codec for remote broadcasting; perfect for studios, TOCs and even remote kits. Z/IP ONE has all the codecs you can think of: AAC-ELD, AAC-HE, AAC-LD, MPEG 4 AAC, MPEG 2 AAC, MPEG Layer 2, G.711, G.722 codecs, plus linear audio and optional aptX® Enhanced coding. Z/IP ONE supports SIP 2.0 protocol and conforms to N/ACIP standards; it also works with VoIP devices and connects to compatible SIP PBXs. And it even comes with a matched USB WiFi stick for wireless operation. So wherever you go, you'll be able to Hear from There with Z/IP ONE.

About Telos

Steve Church founded Telos Systems in 1985. As both a talk show host and radio group Technical Director, Steve was only too familiar with the frustrations of "bad phones" and even less responsive equipment manufacturers, so he set about eliminating the technical problems that plagued radio call-in segments.

In 1984, he invented the Telos 10, the first DSP-based telephone-to-broadcast interface system – allowing radio stations to significantly improve the technical quality of call-in segments. The overwhelming response to Steve's economical and technically elegant solution to a nagging problem provided the spark from which Telos was born.

A lot's happened since then. We pioneered the use of MPEG Layer 3 coding in the revolutionary Zephyr ISDN codec. We produced the first hardware MP3 streaming encoder for broadcast. We developed the world's first "whole-plant" broadcast phone system. And we invented the IP-networked radio console, and then integrated broadcast phones into that network via Ethernet.

Telos has grown steadily since our initial production run of 25 Telos 10 units in 1985! With tens of thousands of systems in the field, it now is hard to find a broadcast facility in the world without at least one piece of our gear. Our organization, now called The Telos Alliance, includes the Omnia Audio, 25-Seven Systems, Axia Audio and Linear Acoustic brands, and our R&D department – the largest research team in broadcasting – continues to develop innovative audio products for radio and television broadcasting, telephony, and the Internet.

Z/IP ONE: THE IP WAY TO HEAR FROM THERE

With Z/IP ONE (the “Z/IP” stands for “Zephyr IP”), you don’t have to compromise audio quality for a solid connection. Z/IP ONE helps you get the best possible quality from public IP networks and mobile data services — even from connections behind NATs and firewalls.

Z/IP ONE Provides Superior Audio Quality

To make certain your remote broadcast has excellent audio quality even when IP connections are not-so-excellent, Telos engineers employed a new codec based on low delay AAC. It’s called AAC-ELD (Advanced Audio Coding-Enhanced Low Delay), and it produces excellent fidelity at low bitrates with nearly inaudible loss concealment and very little delay. Standard high-performance codecs are a part of the Z/IP ONE toolkit as well, such as AAC-HE, AAC-LD, MPEG4 AAC-LC, MPEG2 AAC-LC, G.711, G.722 and even linear PCM. And if apt-X® is part of your codec cache, you can add it to your Z/IP ONE as a small extra-cost option.

Simple, No-Hassle Operation

Z/IP ONE is from Telos, so of course you expect that it will be easy to set up and easy to use. And it is — the front panel controls are intuitive and friendly, and the built-in Web server makes short work of configuration or remote control, using any PC with a Web browser. And our exclusive worldwide Z/IP Server service, free to Z/IP owners, lets you easily get around NATs and network firewalls for fast connections to your favorite locations. For even more flexibility, Z/IP ONE can connect to third-party apps such as LUCI LIVE and LUCI LIVE Lite to receive on-the-go reports from smartphones and tablets.

Plenty Of I/O To Go

On the Z/IP ONE rear panel, you’ll find balanced analog XLR ins and outs, a Livewire+ LAN port for quick connection to Axia networks, and a separate WAN port for safe connection to “the outside world.” If your studio calls for using AES/EBU audio, it’s available optionally at small extra cost, also presented on professional XLR connectors.

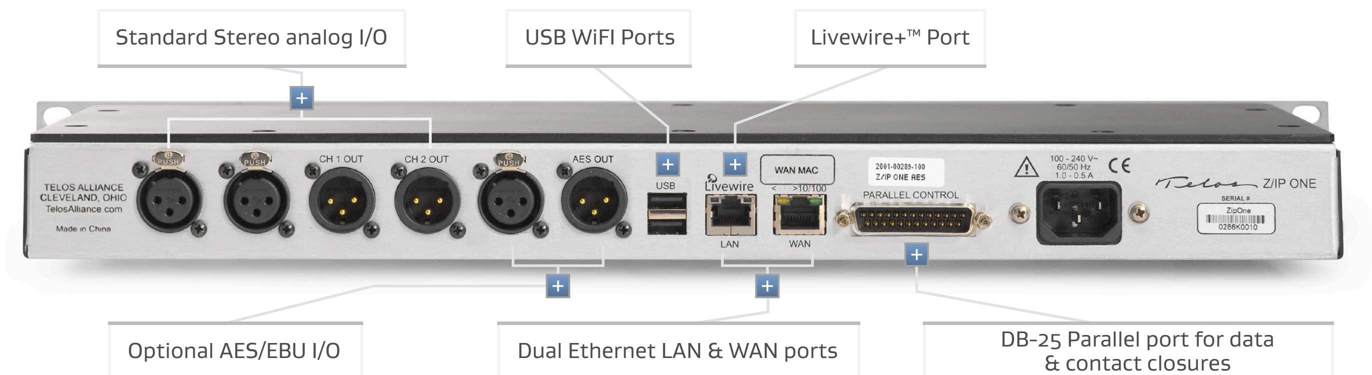
Z/IP ONE is also wireless-capable and connects natively to IP networks via Wi-Fi. A parallel port is provided for end-to-end, time-aligned GPIO contact closures; Z/IP ONE can also transport RS-232 serial data (using an inexpensive USB-to-Serial adaptor cable), synchronized with audio delivery — useful for RDS/RBDS data, as well as other serial data, at up to 9600 bps.

Rock-Solid Connections

We all know the Internet’s not perfect. That’s why Telos collaborated with Fraunhofer (the developers of MP3 and many AAC breakthroughs) to develop a unique coding control algorithm that adapts to changing Internet conditions on the fly, helping you maintain quality and stability.

We call it ACT, short for Agile Connection Technology, and only Telos has it. Using ACT to sense and adapt to the condition of your IP link, Z/IP ONE delivers superb performance on real-world networks. ACT adapts dynamically to minimize the effects of packet loss and jitter. When the bits are flowing smoothly, you’ll benefit from the lowest possible delay and the highest possible fidelity. If congestion starts to occur, Z/IP ONE automatically lowers bit rate and increases buffer length to keep audio flowing at maximum quality. You’ll get reliable audio even when network conditions are unpredictable — and you won’t need to fiddle with settings or codecs to do it.

FEATURES AND BENEFITS



Slim, light, efficient. Z/IP ONE is designed to take up the least space possible in your studio, TOC or remote kit. Z/IP ONE weighs less than 5 pounds and occupies only 1RU of rack space. And it draws less than 15 Watts of power – no cooling fans or “wall wart” power supplies here.

Easy, intuitive controls. Z/IP ONE’s clear, bright OLED front panel has an intuitive menu design that makes it easy to quickly find and use the tools you need. Just rotate the adjacent knob to highlight the option you want, and push to select. Top-level menu provides direct access to frequently-used Phonebook and auto-dial settings, as well as the Setup Wizard and on-screen help. When connected, send/receive meters and connection quality meter are displayed.

Front Panel Dialing. Some codecs make you dial connections using a PC interface — not very intuitive. Z/IP ONE puts its dialing controls right on the front panel, where they belong. **AUTO** button takes you right to your Phonebook for quick connection to saved locations; **CONN** key lets you dial on-the-fly, and pressing **DISC** *twice* ends the call and readies you for your next connection.

Built-in headphone port. Separate headphone amp at a remote? Not with Z/IP ONE. Plug your 'phones into the convenient front-panel jack. Turn the knob to set volume; push it to switch the headphone feed between Send and Receive audio. Sweet!

No wall-warts here. Telos gear uses telecom-grade internal power supplies, exclusively — no cheap external plug-ins to burn up or lose. Cool-running, auto-ranging supply with a standard IEC receptacle works anywhere in the world. Z/IP ONE is power-efficient, too — only draws 14.2 Watts.

AES67 Livewire+

Dual Ethernet ports. Every Z/IP ONE comes standard with Livewire+™ I/O, for easy, one-cable Ethernet audio and data connection to Axia AoIP networks. And to keep your studio network safe, there's a separate, firewalled WAN port for connecting your Z/IP ONE to the outside world.

Choose your I/O. In addition to Livewire+ I/O, every Z/IP ONE has stereo analog ins and outs, presented on professional, balanced XLR connectors. Want AES/EBU connections? No problem — AES I/O is available at a modest extra cost.

Data On Demand. The DB25 connector on the back panel provides contact closures (close-to-ground) inputs and outputs. There are 8 configurable open-collector outputs, and 8 open-collector inputs that provide time-aligned RS-232 data for remote control or RDS metadata, as well as GPIO signaling and control.

Go Wireless. No hard line? No problem. USB ports on Z/IP ONE rear panel let you connect via WiFi, with the matching USB WiFi stick included with every unit. WiFi, WLAN and UMTS/EVDO networks are supported.

Z/IP Server Connects You. NATS and firewalls are essential for security, but hard to navigate. Telos' complimentary, worldwide Z/IP Server service makes it easy to connect, by keeping track of every Z/IP ONE that's online, enabling you to traverse corporate firewalls and NAT layers and get on the air with just one click.

Smart(phone) Connections. Z/IP ONE connects to third-party apps, like LUCI LIVE and LUCI LIVE Lite. Your talent can call in with instant, on-the-go reports from their smartphone or tablet, whenever and from wherever the action is breaking — with stunning audio quality.

FAQS

What are the major differences between an Analog Z/IP ONE and the Z/IP ONE with AES?

The AES/EBU design has a different main board with a digital audio controller. Differences between the two units include:

- Different analog input gain settings, particularly with respect ,to microphone gain,
- Two additional XLRs on the back panel of the AES/EBU unit, and
- A ground stud near the power entry module on the back panel of the AES/EBU unit.

Does the Z/IP ONE with AES still support analog audio?

Yes. Analog inputs and outputs are still available with configurable gain, and even an available microphone boost setting.

Can an analog Z/IP ONE be upgraded to use AES/EBU?

No, sorry. This is a factory-only option and is not field-installable.

Can I use AES and analog audio at the same time?

Yes. Since the Z/IP ONE encodes just one stereo pair, you must choose which of the input options to use for encoding. However, decoded audio is simultaneously available on Analog, AES/EBU, and Livewire+ outputs. You can even set the inputs to 'fail over' from Livewire+ to AES/EBU to Analog in case of input failures.

What determines an input failure?

Loss of frame synchronization is used to determine whether to change input sources from Livewire+ or AES/EBU. Silence is not considered.

I've never heard of N/ACIP compliance before. Why is that so important?

Telos would love it if everyone purchased a Z/IP ONE codec! But there are other brands available, and the N/ACIP standard assures interoperability between different brands. This means that your Z/IP ONE supports G.711, G.722 and MPEG-1/2 Layer II, plus PCM, and can easily work with the other guys' codecs.

I've still got my original Zephyr XStream in the rack. Will it work with Z/IP ONE?

Of course. Z/IP ONE is backward compatible with thousands of Telos Zephyr Xstreams in the field.

I can envision our cluster using a number of Z/IP ONEs. Is there any way to logically organize them all so they can talk to each other, or the outside world?

Yes, Z/IP ONEs can be organized into groups. Groups make it easier for others to find you, and simplify the task of selecting a name for the device. For security, groups can be password protected, to prevent folks outside your organization from viewing the names of the Z/IP ONE units within your group. Of course, you can also create a list to allow selected people to access your group.

By design, every Z/IP ONE is initially configured to be a member of the "public" group. Beyond that, groups can be created at any time, and you may create as many groups as you need. Each Z/IP ONE may be assigned a name that is unique within your group.

You mentioned that Telos has something called a "Z/IP Server". Tell me more?

We have a Z/IP Server service that's available, free, to Z/IP ONE owners worldwide. It provides a set of utilities that enhance the functionality of your Z/IP: a directory service, a presence service, NAT traversal service, and media relay service. Individual stations and organizations can also host their own, private Z/IP Servers if they like. Download the manual if you'd like to learn how to set one up.

What's this "presence service" on the Z/IP Server that you mentioned?

The "presence service" allows you to see the state of the other Z/IP ONE units in your "buddy" (or speed dial) list. This way, you know if the Z/IP ONE you want to connect to is online, offline or busy — before you even attempt the call! The status information is updated a few times per minute to keep the list up to date.

The Z/IP Server sounds really cool. But is there a way to make some of my Z/IP ONEs public, and keep the others restricted?

Yes! Isn't it nice not having to remember all those pesky IP addresses? When you register with the Telos Z/IP server, your Z/IP ONE can select one the following visibility modes:

1. Visible to all: Anyone is able to see your Z/IP's directory listing.
2. Visible to group: Only visible to those devices that know your group's password. Of course, this option is not useful for the public group since anyone knows the public group's password.
3. Hidden: Your Z/IP is not shown in the server directory. Others are still able to call you if they know your device's name and group.

FAQS

What exactly is a NAT?

Network Address Translation (NAT) is a technique that allows multiple IP devices on a private network to share a single public IP address. A NAT device has two network ports, one for the "private side" (the LAN) and the other for the "public side" (the Internet). As the name suggests, the NAT router translates the internal IP addresses to the public IP address.

How does the Z/IP Server help with NATs?

Making NAT configuration changes is a pain. Your Z/IP ONE uses the Z/IP Server to detect when it is installed behind a NAT, and then determine the type of NAT. It is then able to accept calls from the outside without requiring you to make special configuration changes to your NAT. In addition to NAT traversal assistance, the Z/IP Server also allows you to see the status of your "buddy" devices, to get a directory listing of available devices, or to dial another device by name.

Are all NATs the same?

Sadly, no. There is no defined standard on how a NAT should behave. Each vendor implements the NAT functions in a proprietary manner. You may hear NATs described as "full cone", "restricted cone", "port restricted cone" or "symmetric". Not everyone agrees on the exact definition of these terms, and to date, there is no universal standard.

How do NAT devices affect IP-Audio communication?

Well, there's good news and bad news with NATs. The good news: NATs are a lower cost solution compared to obtaining multiple public addresses. They also add to security by hiding the internal structure of the network, and by allowing only pre-specified direct contact via port forwarding (see below).

The bad news: internal devices are no longer reachable from the outside because the NAT does not know which internal device the packet is destined for. To get around this drawback, NATs have configuration capabilities which allow the user to forward specific ports to a given internal device or to add a single device DMZ (De-Militarized Zone), where this device is now visible to the public.

This is why Telos' complimentary Z/IP Server service is so useful: using Z/IP Server alleviates the trouble of manually configuring your device to route around NATs, by doing it for you.

SPECIFICATIONS

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.
- Optional: apt-X Enhanced ® from CSR.

CONNECTIONS:

ANALOG

- 1x Stereo input, presented on two XLR-F connections
- 1x Stereo output, presented on two XLR-M connections

LIVEWIRE+

- 1x 100Base-T connections, presented on RJ-45

AES/EBU (OPTIONAL)

- 1x Stereo Input, presented on one XLR-F connection
- 1x Stereo Output, presented on one XLR-M connection

NETWORK

- 2x 100Base-T connections, presented on RJ-45 (1x LAN, 1x WAN)

USB

- 2x A-Type, Female

PARALLEL (GPIO)

- 1x DB25, Male

AUDIO:

ANALOG LINE INPUTS:

- Input Impedance: 6K Ohm differential
- Input Range: Selectable, Line (+4 dBu nominal), Microphone (-50dBu nominal)
- Selectable Phantom power

ANALOG LINE OUTPUTS:

- Output Impedance: 50 Ohm differential
- Output Clipping: +22dBu

SPECIFICATIONS

DIGITAL AUDIO INPUTS AND OUTPUTS

- Reference Level: +4 dBu (-20 dB FSD)
- Impedance: 110 Ohm, balanced
- Signal Format: AES3 (AES/EBU)
- AES3 Input Compliance: 24-bit with sample rate conversion
- AES3 Output Compliance: 24-bit
- Digital Reference: Internal (network timebase) or external reference 48 kHz, +/- 2 ppm
- Internal Sampling Rate: 48 kHz
- Input Sample Rate: 32 kHz to 192 kHz
- Output Sample Rate: 48, 44.1 or 32 kHz, or "sync to input" (auto-matches rate and clock from AES/EBU input)
- A/D Conversions: 24-bit, Delta-Sigma, 256x oversampling
- D/A Conversions: 24-bit, Delta-Sigma, 256x oversampling

FREQUENCY RESPONSE

- Any input to any output: +/- 1dB 25– 20 kHz

HEADROOM

- 18 dB

ARCHITECT SPECIFICATIONS

DYNAMIC RANGE

- 87dB Unweighted
- 90 dB "A" Weighted

TOTAL HARMONIC DISTORTION + NOISE

- < 0.03% @ +12dBu, 1 kHz Sine

CROSSTALK ISOLATION

- > 80 dB

POWER SUPPLY AC INPUT

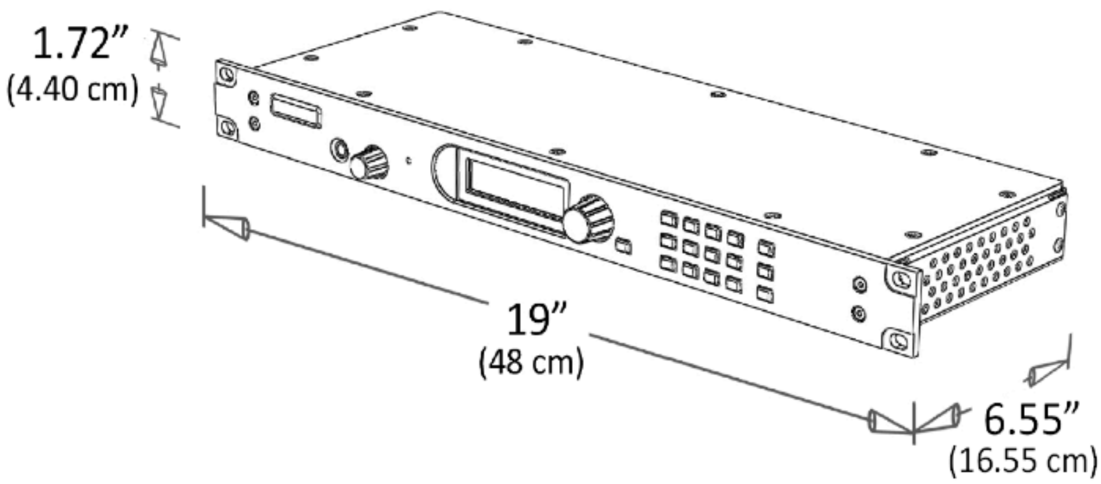
- Auto-ranging supply, 90VAC to 240VAC, 50 Hz to 60 Hz, IEC receptacle, internal fuse
- Power consumption: 14.2 Watts

OPERATING TEMPERATURES

- 0-40 degrees C (32-104 degrees F), stirred air

DIMENSIONS

- 19" (48.3 cm) standard rack mounting front panel
- 1.75" (4.5 cm) height, 6.5" (16.51 cm) depth
- Shipping Weight: 8 lbs. (3.62 kg)
- Shipping Dimensions: 24" x 14" x 6" (61 cm x 35.6 cm x 15.25 cm)



ARCHITECT SPECIFICATIONS

Specification Overview

The following describes an appliance, normally used in a pair in diverse locations, for transporting high-quality, stereo audio over private (WAN) and public (Internet) IP connections. The unit shall operate in several modes and with several stream setup protocols, as well as offer both contemporary and legacy audio codecs for maximum flexibility and compatibility with other IP codecs. The device shall be fully compliant with the N/ACIP standard, established by the EBU <http://tech.ebu.ch/docs/tech/tech3326.pdf>, which describes inter-connectivity of IP-connected audio codecs.

The unit shall additionally offer connectivity, convenience, and performance far in excess of the N/ACIP standard. The unit shall incorporate numerous features and services affording a simple and automatic approach to connecting with other identical units, as well as comprehensive configuration and operation choices to obtain optimal audio performance over typical IP links.

The device shall be of a professional design, suitable for use by broadcasters, net-casters, content creators, news reporters and bureaus, and others requiring either instant, convenient, and simple audio connectivity with other units, or those needing reliable, 24/7 unattended operation. The appliance shall feature professional audio inputs and outputs and a selection of popular, state-of-the-art audio coding algorithms at a wide selection of bit rates.

The unit will also offer convenient front-panel monitoring and basic configuration, browser-based remote monitoring and configuration, and effective GPIO-based remote control and event monitoring.

The device will afford audio, connection and GPIO status, and streaming confidence monitoring via a front-panel headphone connection, rear-panel professional outputs, and a self-contained HTTP server.

Through its design philosophy and best practices in a contemporary IP network environment, this device shall be suitable for short-term voice-over and remote broadcasts, long-form program feeds, remote talent connection, full-time 24/7 audio links such as STL, and world-wide inter-city relay applications.

Physical

The appliance shall consist of a 1RU, 19" standard rack-mount enclosure. The front panel shall be an attractive, functional design with a bright, clear OLED-based display, combination navigation/selection rotary encoder, "back" or "ESC" button, combination dialing/alphanumeric keypad, dedicated AUTO, CONNect, and DISConnect buttons, headphone jack, and headphone volume control. The appliance shall be silent in operation, fanless, and connect to AC power via an IEC power entry module. AC power input shall be of universal design, accommodating worldwide standard AC power voltages and frequencies.

The rear panel shall provide analog audio inputs via two XLR-F bulkhead connectors for Left and Right audio, and analog outputs on two XLR-M bulkhead connectors for Left and Right audio. All audio inputs, outputs and streams shall be stereo. The rear panel shall provide two Ethernet/IP (network) connections on standard RJ-45 bulkhead connectors. Though flexible for alternate configurations, the RJ-45 Ethernet connectors shall be labeled "WAN" and "Livewire+ LAN". The Livewire+ LAN port may be used to connect with a Livewire+ AoIP network, including AES I/O devices.

Audio, GPIO, and Data Input and Output

The appliance shall offer analog and Livewire+ audio input and output with software-selectable audio I/O levels to accommodate professional broadcast equipment connections. The analog audio input, presented via bulkhead XLR-F connectors, shall be selectable for professional line level (+4 dBu) or mic level (-50 dBu, adjustable). When selected as mic level inputs there shall be an option to supply phantom power for condenser mics.

The AoIP audio I/O functionality (Livewire+ RJ-45 Ethernet port) shall be fully compatible with the Livewire+ AoIP standard, including compatibility with iProbe and Pathfinder functions.

The device shall offer two USB ports on the rear panel. These ports shall support an approved WiFi radio and a USB to RS-232 Serial Data adapter. An approved WiFi radio shall be included with the product, and the user interface shall support its use in specifying and connecting to WiFi Access Points whether unencrypted or secured, using WEP, WPA, or WPA2 encryption protocols. An optional USB to RS-232 Serial Data adapter shall be available from Telos.

ARCHITECT SPECIFICATIONS

The device shall offer GPIO connectivity which is flexible and user-configurable. Eight Input and eight output pins shall be available on a rear-panel D-sub 25 male connector. These pins may be individually configured for several functions as needed by the user. These functions shall include end-to-end contact closures or single-ended inputs and outputs to perform or indicate status of the local device. Outputs (GPOs) may be configured to indicate the following local device states: Connected, Receive Lock, Follow Input, Disrupted Call, Lost Packet, Bad Quality (threshold settable). Inputs (GPIs) may be configured to effect the following local device functions: Disconnect, Dial (any Autodial entry), Send constant 1 or 0.

The device shall include the ability, when optionally equipped, to communicate RS-232 Serial Data bi-directionally with the far-end device at a rate of up to 9600 kbps. This capability shall apply when using an MPEG codec as this ancillary data is embedded within the audio data frames.

Both GPIO messages and serial data shall be synchronized with the attendant audio encoding/decoding process such that real-time switching of far-end equipment may occur in synchronization with the audio being delivered in each direction.

Encoding and Transport

The appliance shall perform audio bit-rate-reduction (encoding and decoding) via a selection of industry-standard, psychoacoustic algorithms. Specifically, the appliance shall offer AAC-ELD, AAC-HE, AAC, AAC-LD, MPEG2-AAC (for Telos Extreme compatibility), MPEG Layer II, G.722 mono, G.711-uLaw mono, plus 16, 20, and 24 bit PCM Stereo audio encoding/decoding algorithms. The appliance shall offer target bit rates from 18 kbps to 320 kbps, depending upon the algorithm selected, plus 1.5 to 2 Mbps for PCM (linear) modes.

Operational Modes

The device shall offer several operational modes, selectable by the user to support the application. These modes of connection and operation include the following: **1)** Convenient server-assisted "dialing" by selecting a friendly name of a destination device from a customizable list; **2)** Direct "dialing" of another similar device without using an outside server; **3)** "Push dialing" of another device, establishing a one-way audio connection, from caller to destination; **4)** "Push dialing" up to 4 devices, establishing one-way audio toward the destinations.

Agile Connection Technology

The device shall offer a method for automatic and ongoing negotiation of encoder bit-rate and decoder buffer size at each end, within user-definable parameters. Such functionality will, within limits, adjust to moment-by-moment changes in IP connection available bandwidth and latency. This technology shall allow the user to maintain a two-way audio and data connection with the lowest possible delay at all times, enabling effective, two-way conversations and cueing over jittery IP networks, including the Public Internet.

Rendezvous Server

The device shall support communication with a "rendezvous server", enabling NAT Firewall Traversal and presence status monitoring of approved similar devices. One or more Rendezvous Servers shall be operated and maintained by the manufacturer to provide Directory, Connection, and NAT Firewall Traversal services. The rendezvous server(s), working with complimentary technology in the device, shall allow for convenient connection with other similar devices by non-technical users.

Network Support

The appliance shall support two self-contained Ethernet/IP network interfaces. Each network connection operates independently, each having its own MAC address, and may be individually assigned an IP address and other standard networking parameters. Either connection may be used for remote control, monitoring, and streaming. One of the connectors may also be used for AoIP (Livewire+ standard) audio I/O.

System Updates

The appliance will allow two versions (banks) of operating software to be stored internally. The appliance shall support the updating of its operating software via standard web browser access and file upload. The appliance shall further support web or file-based software updates with selection of booting to either software bank.

Support and Warranty

The appliance shall be offered with a standard limited warranty period of five years. English-language factory support shall be available to users at no charge on a 24/7 basis.

THE TELOS ALLIANCE

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