HyperCloud series Voice to Cloud Streamers



The OptiLogix HyperCloud OEM Voice to Cloud Streamers provide powerful features for building advanced cloud based Call Recorders.

Each unit combines a compact form factor with low power consumption resulting in unmatched reliability.

Fully stand-alone embedded ARM based Linux system with DSP technology for D-channel decoding, voice processing / encoding and IP streaming.

D-channel signalling supports Call Setup, Connect, Clear, DDI number and CLI number decoding.

Supports all major ISDN variants, Q.SIG, DASS-2 and DPNSS.

Audio file buffering and low bandwidth requirements when streaming / uploading to the cloud using the integrated high quality low bitrate audio encoders.

Features and Benefits

Compact box with telecoms interfaces, 10/100 Mbit/s Ethernet port and LCD display

Non intrusive and undetectable high impedance passive monitoring

Dialled number and Caller ID signalling support

Models available for PRI-ISDN, BRI-ISDN / S_o bus, Digital handsets and Analog handset / trunk lines

Analog models support DTMF, FSK Caller ID, AGC and audio detection

Digital models support all major PBX with highly accurate DigitalVox start / stop triggering

Web interface for system configuration, firmware uploading and channel capacity license upgrading

Fully stand-alone embedded (Linux) operation with internal RAM and (expandable) SSD storage for audio file buffering.

Supports 64kbit/s A-law, 16kbit/s MP3, 13kbit/s GSM and 8kbit/s MP3 high quality audio encoding for reduced IP bandwidth cloud streaming

Optional highly secure encrypted audio files.

FTP and Secure FTP for streaming audio files and metadata to the cloud.

CE, FCC and RoHS2 compliance



Technical Specifications

Mechanical characteristics: Operating temperature: Humidity: Maximum power requirements: Embedded operating system:

Interface Specifications

Primary Rate interface: AC impedance: Maximum tap length: Protocols:

Basic Rate interface: AC impedance: Maximum tap length: Protocols:

Digital handset interface: AC impedance: Maximum tap length: Protocols:

Analog handset / trunk interface: DC/AC impedance: Maximum tap length: Signalling:

Audio Processing

Voice and Silence detection:ProgrammableUpstream and downstream audio gain:ProgrammableFrequency response:300-3400Hz (alSpeech encoding/compression:64kbit/s A-law,

Programmable Programmable 300-3400Hz (all compression modes) 64kbit/s A-law, 16kbit/s MP3, 13kbit/s GSM and 8kbit/s MP3

Safety and EMI Certifications

Safety, emissions, immunity: Compliance: Estimated MTBF: Warranty: EN 60950, EN 55022, EN 55024 CE, FCC and RoHS2 500.000 hours 3 years

The OptiLogix policy is one of continuous development and consequently the equipment may vary in detail from the description and specification in this publication



www.optilogix.com

Compact box with external power supply adaptor 0 °C to +50°C 5% to 80% non-condensing +12Vdc (1000 mA) Linux

E1 (2.048Mbit/s) , T1 (1.544Mbit/s) 1100 Ω 10 m (unterminated), 100 m (terminated) All major ISDN variants, Q.SIG, DASS-2 and DPNSS

 $\begin{array}{l} 4 \text{ wire } S_o \text{ bus} \\ \text{Line Matched} \\ 500 \text{ m} \\ \text{Euro-ISDN} \end{array}$

2 wire bus Line Matched 500 m All major PBX supported (DigitalVox triggering)

2 wire voltage start or line level audio triggering (Vox) Infinite / 3000Ω 5000 m Ring detection, voltage detection, DTMF detection for dialled numbers, FSK Caller ID detection, voice activity detection