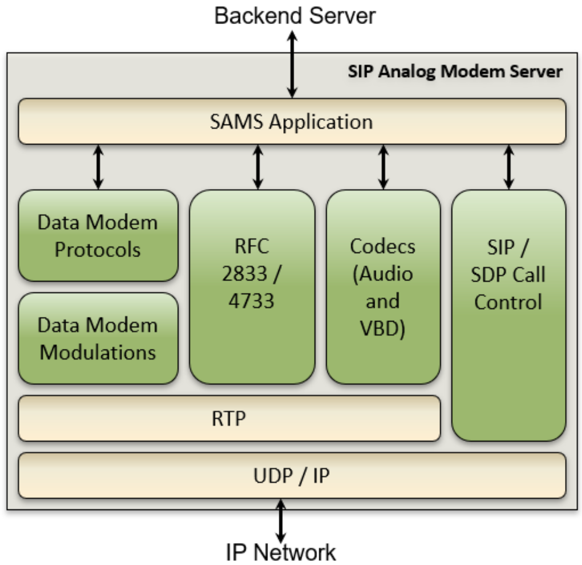
SIP Analog Modem Server (SAMS)

The VOCAL SAMS is a solution for interoperating with legacy modems without the need for modem banks with E1/T1/ISDN connections.

As E1/T1/ISDN lines are increasingly being replaced by Ethernet based VoIP systems, companies are often left with modem based technology that can no longer work with the new infrastructure, but remain necessary for daily business. This can happen with servers interacting with point-of-sale devices, meters, alarm systems and numerous industrial controllers, VOCAL has a number of solutions to solve these problems.



SAMS software enables legacy modems to communicate over IP networks

VOCAL's Virtual Modem Server allows a business to deploy a server in the cloud, which can work with the modern VoIP phone network infrastructure, based on SIP and RTP, instead of necessitating T1/E1/ISDN lines to connect to the PSTN. The VOCAL Virtual Modem server contains true soft-modems, not just an AT command set on a TCP socket, pretending to be a modem. This allows the server to connect to true voice-band modems deployed on the PSTN, and thus does not require installation of additional equipment on the far end of the system. The server is controlled by industry standard AT commands via standard TTY nodes in the server's filesystem – just as if it were a box-modem on a serial connection. This allows it to easily fit into currently existing systems. Many other configurations and methods of control are available in order to flexibly meet the needs of already deployed infrastructure and applications. In addition, VOCAL can customize or build to suit, in order to meet very specific system requirements.

Modem over IP (MoIP) solutions treat modem audio signals as simple data, which may be packetized and transmitted in real-time over IP networks. Audio tones are sampled and digitized at (or near) the source, the data packets are transmitted over IP networks using VoIP protocols, and demodulated at the destination. As such, Modem over IP eliminates POTS lines and associated expenses. However it does not eliminate the redundancy of encoding and decoding modem audio at the source and destination nor the modem banks at the ASP.

The Software is part of a fully integrated and highly configurable VoIP software solution with a Network Stack, SIP Stack, secure communications, full-featured Telephony software, and a comprehensive data modulation and data modem software library.

Features

- 100% software solution easily migrates to the cloud
- Runs on any standard Linux distribution
- Replaces RAS equipment configurations when analog lines are being replaced with VoIP (SIP/RTP) lines
- Scalable – enables a large number of modems on a single server (10s to 100s depending on the server and modem rates)
- Exports TTY (com port) interface for each modem
- Each TTY accepts industry standard AT commands - V.250, Industry Standard
- Registers with one or more SIP servers (can be phone number per line, modem pool hunt, or other combinations)
- SRTP and Secure SIP options available
- Dialing commands will make outgoing SIP/G.711 call (with modulation data in the stream)
- Incoming SIP calls will report RING to applications listening to TTYs
- Compatible with ppp and pppd etc.
- Requires no analog audio interface
- Slaves to far end modem clock to eliminate skew
- High speed data modulation - V.34, V.90, V.92 (optional)
- Low speed data modulation - V.32, V.32bis, V.22, V.22bis, V.23, V.21
- Bell modulation - Bell 212, Bell 103
- Supports standard protocols such as V.42 / LAPM, MNP 2-4
- Supports standard compression protocols such as V.42bis and MNP5