SIP Encrypt

A Session Initiation Protocol (SIP) Trunk links a business phone system (PABX) directly to the Public Switched Telephone Network (PSTN) using an IP based connection.

SIP Encrypt takes things one step further, providing a SIP Trunk service that is fully encrypted between the customer premise and the Firstcom network. It's a great replacement for ISDN30 as it is a fully managed service when using our connectivity, meaning any issues or problems are visible to Firstcom, usually ahead of a customer noticing. Unlike widely available SIP trunk services from many suppliers, SIP Encrypt introduces true security while retaining the flexibility that SIP offers.

Security

The SIP Encrypt service uses the Advanced Encryption Standard 256 algorithm, which is the prevailing benchmark across the world after being initially adopted by the US National Security Agency. Hackers will recognise the challenge of breaking in should they gain access to the transmitted data, and simply move on to easier targets.

Note that SIP Encrypt is an important asset when companies review their security policies as part of the upcoming General Data Protection Regulation (GDPR), which takes effect in May 2018. It enables organisations to demonstrate material steps toward securing voice calls that may contain personal data about employees or customers.

Deployment Options

SIP Encrypt can be implemented in 3 different ways to suit the type of on-premise communication being used:

1. If the IP PABX supports Transport Layer Security and SRTP it can be configured by the maintainer to receive SIP Encrypt service directly.
2. Safe Gateway will be deployed on site in front of an existing PABX that does not support Transport Layer Security and SRTP or does not require any SIP header manipulation.
3. Firstcom offers a Cisco CUBE SBC when SIP header manipulation is required and the PABX does not support TLS and SRTP

The deployments above may be over Firstcom supplied circuits (Broadband, Leased Line, Converged voice & data), which will mean a fully managed service is delivered – including the remote monitoring of service performance. If SIP Encrypt is deployed over 3rd party circuits then the SIP calls remain securely encrypted but there is no remote monitoring as the circuits are managed by another carrier.

All the usual SIP benefits

The SIP Encrypt service retains all the main benefits offered by SIP Trunks, including scalability, speed of deployment, flexibility and cost competitive versus traditional ISDN30 and ISDN2 offerings.

Security features	Notes
Encryption	AES 256 algorithm, as approved by the US National Security Agency
Extra security	Uses Username versus IP address for authentication so less prone to fraud
Safe Gateway	Suitable when the PBX doesn't support TLS and SRTP and no SIP header manipulation is needed.
Cisco CUBE	Suitable when the PBX doesn't support TLS and SRTP and SIP header manipulation is needed.
Direct connection to IP PABX	Safe Gateway or SBC not needed if the PABX supports Transport Layer Security and SRTP
Credit cap	We will consult with the end customer on a suitable credit cap to help avoid any unforeseen charges that may be fraudulent

Service features	Notes
Fully managed service	Available when using Firstcom supplied access circuit: Broadband, Leased Line or Converged Voice and Data circuit
Works over existing, non-Firstcom circuit	SIP Encrypt will provide the fully encrypted service, however Firstcom will not have visibility of any circuit issues preventing proactive support
Phone number flexibility	A number with a new area code can be selected when using SIP, or an organisation may keep their original number. Firstcom facilitates number porting so you can bring your number to use our services.
Resilience and number diversion	Should an organisation need to re-direct incoming calls, they can do so easily, making local emergencies e.g. fire alarms, easier to handle
Voice Codecs	G.711, G.729 supported and others on application
Fax	T.38 fax supported across the network
Calling Line Identity Presentation (CLIP)	Incoming calls present the available number details of the calling party
Incoming call 'forking'	Calls to a number can be presented to multiple end points (locations) simultaneously. Ideal for meeting answer time SLAs.
PABX compatibility	All leading business phone systems now support SIP, and Firstcom has proven interoperability with these - including Asterisk based systems