SIPREC / SIP
COMPATIBILITY GUIDE

TelStrat's Engage Contact Center Suite provides business and contact center users the affordable, flexible tools they need for call recording, workstation screen capture, quality management, desktop anaytics, speech analytics and/or workforce management. Engage Suite is compatible with a wide range of PBX platforms, telephony technologies, and flexible deployment environments.

Product Features:

- Full call recording automatically, according to user-defined rules, or on-demand.
- Live monitoring of calls & desktop activity for one or multiple simultaneous stations until monitoring session is closed.
- Recorded audio files can be playbacked, downloaded as .MP3 or .WAV, emailed as a file attachment or playable URL, or even played back directly from a customer's CRM application.
- All call information is stored and searchable. Add user-defined fields to call records from 3rd-party CRM applications, such as customer ID or policy number.
- PCI-DSS, HIPAA, FIPS, and other regulatory compliance program requirements with auto pause/resume.

Technical Capabilities:

- Multi-tenant capabilities for easy administration and centralized management of multiple tenants from a single logon.
- High availability solutions supporting virtual server or physical server deployments.
- Scalable solutions to over 10,000 endpoints.
- Rock-solid security and recording integrity with watermarked audio files, SSL/HTTPS Web access, and optional AES 256-bit encryption.
- Archiving solutions supports SAN, NAS, and attached storage.
- Web services integration supports flow-through provisioning, call notifications, call download, call annotations, and more. Sample applications, source code, and executable files provided.
- On-Premise or Cloud deployments available.
- Cloud subscription deployment options include service providers' data center, TelStrat's data center, or hybrid on-premise/cloud

SUPPORTED INTEGRATIONS

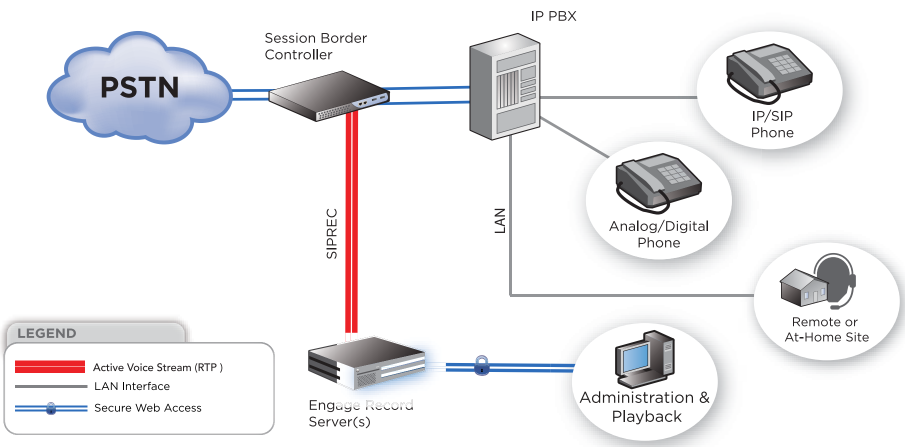
- Any PBX or Call Manager equipped with SIP trunks
- SIPREC
- Any Session Border Controller device
- Any SIP end point

Network Architecture Details

When recording phones in any SIP environment, Engage Record can interface to the PBX with the following recording methods:

SIPREC Trunk Recording

SIP Trunk calls can be recorded using Session Recording Protocol (SIPREC) by deploying a SIPREC capable Session Border Controller (SBC) in between the Public Switched Telephone Network (PSTN) and the IP Private Branch Exchange (PBX). The SBC can be programmed to send a SIPREC invite for a range of extensions to one or more Engage Recording server(s). Engage can be programmed to record all calls or selectively record based on configurable recording criteria.

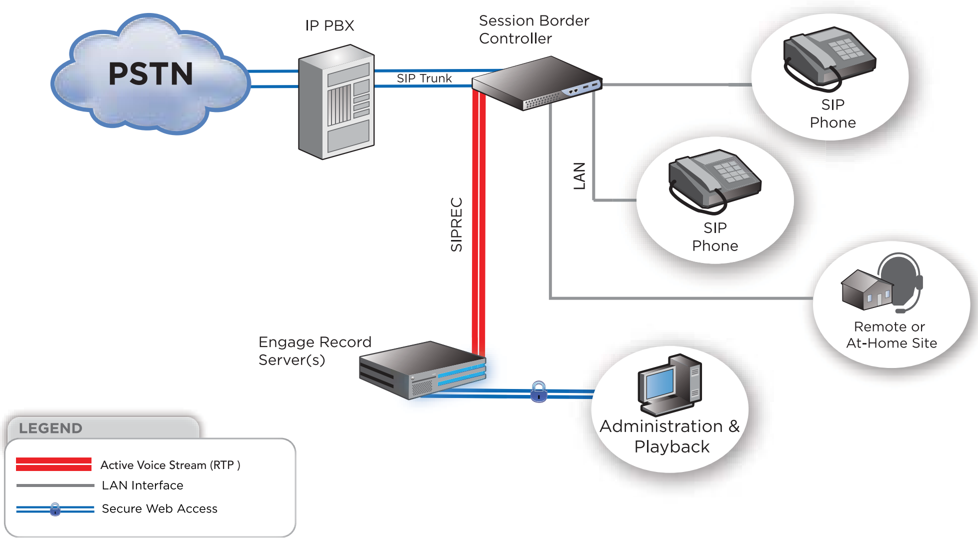


When a recording session is invoked, the SBC delivers the SIP signaling and active media stream to Engage Record through a SIPREC interface. Call event data such as call start, call end, dialed digits, calling line ID, etc is captured from the SIPREC interface. Agent ID is not currently available.

SIPREC Endpoint Recording

SIP endpoints can be recorded using SIPREC by deploying a SIPREC capable SBC in between the IP PBX and the SIP endpoints to be recorded. The SBC can be programmed to send a SIPREC invite for a range of extensions to one or more Engage Recording server(s). Engage can be programmed to record all calls or selectively record based on configurable recording criteria.

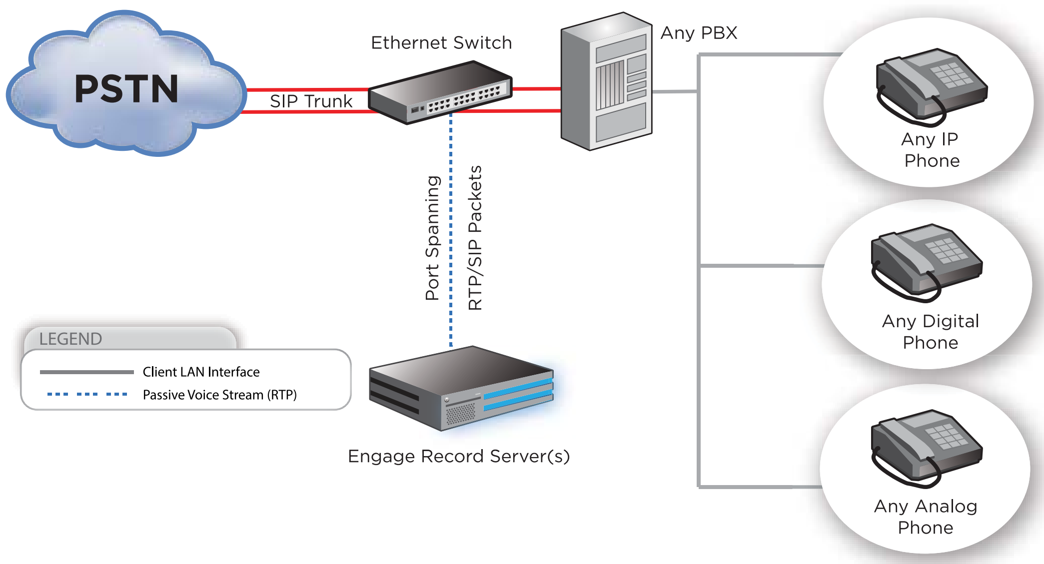
When a recording session is invoked, the SBC delivers the SIP signaling and active media stream to Engage Record through a SIPREC interface. Call event data such as call start, call end, dialed digits, calling line ID, etc is captured from the SIPREC interface. Agent ID is not currently available.



SIP Trunk Port Spanning

SIP trunk calls can be recorded using port spanning. All calls to be recorded are “spanned” to a single contact point on the network where the Engage Record Server connects. A second Network Interface Card (NIC) in the Engage Server collects the spanned Voice over IP (VoIP) traffic to be recorded.

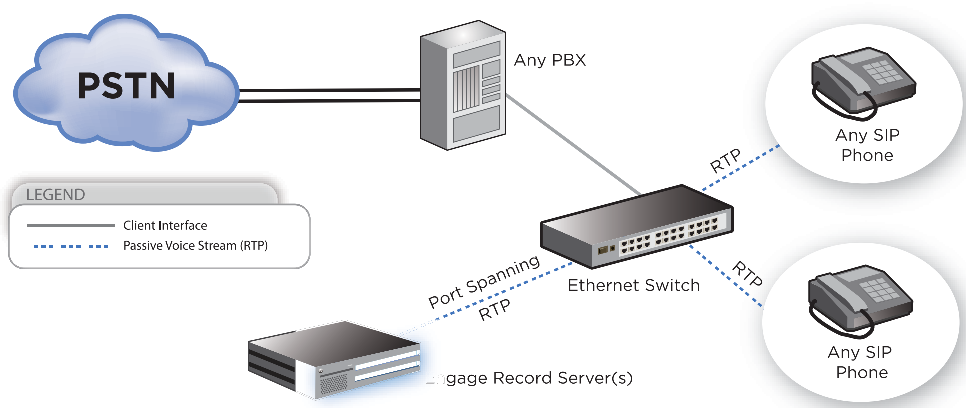
Engage supports any current universal standards-based VoIP PBX or SIP Proxy. PBX features from SIP lines such as multiple extensions per telephone, call conferencing, call hold, call transfer, etc are also supported. Call event data such as call start, call end, dialed digits, calling line ID, etc is captured from the SIP Packet. Agent ID is not currently available.



SIP Endpoint Port Spanning

SIP endpoints can be recorded using port spanning. All phones to be recorded are “spanned” to a single contact point on the network where the Engage Record Server connects. A second NIC in the Engage Server collects the spanned VoIP traffic to be recorded. Screen capture is supported based on the SIP URI of the recorded endpoint.

PBX features from SIP lines such as multiple DNs per telephone, call conferencing, call hold, call transfer, etc. are also supported. Call event data such as call start, call end, dialed digits, CLID, etc is captured from the SIP Packet. Agent ID is not currently available.



DETAILS

Engage Server Requirements:

- **Windows Server 2012 and 2008 (32-bit or R2)** on Engage is supported without limitation.
- **Microsoft SQL Server 2012 or 2008** database applications are supported.
- Optional **RAID 1, 5 or 10** configured internal hard drive, which is recommended for resiliency.
- **Two (2) NIC ports** are recommended to separate the voice network from the data network.
- If using any SIPREC Recording method, any **SIPREC capable Session Border Controller** device is required with sufficient SIPREC licenses.
- If using any Port Spanning Recording method, **Layer 2 Ethernet switch(es)** with switch port analyzer.

A simple installation of Engage Suite will have call recording implemented in as little as one (1) day. With additional support for SIP, VoIP, TDM, analog, and radio voice technologies; customers migrating between PBX platforms can record multiple voice technologies or platforms on a single server simultaneously.

Security Features

Engage Suite secures all Web-based data & communication using Secure Sockets Layer (SSL)/ HTTPS. Microsoft® Single Sign-On (SSO) and Active Directory integration provide robust user access control to satisfy corporate security requirements, with no need to logon multiple times.

To comply with regulatory security requirements, audio files can be watermarked to ensure the recording's authenticity and the included SDK allows third-party integration to stop/start recordings, protecting sensitive data. An available, affordable security package provides end-to-end encryption for all recorded calls and screen captures.

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