SM7000

VoIP Gateway

KEY FEATURES

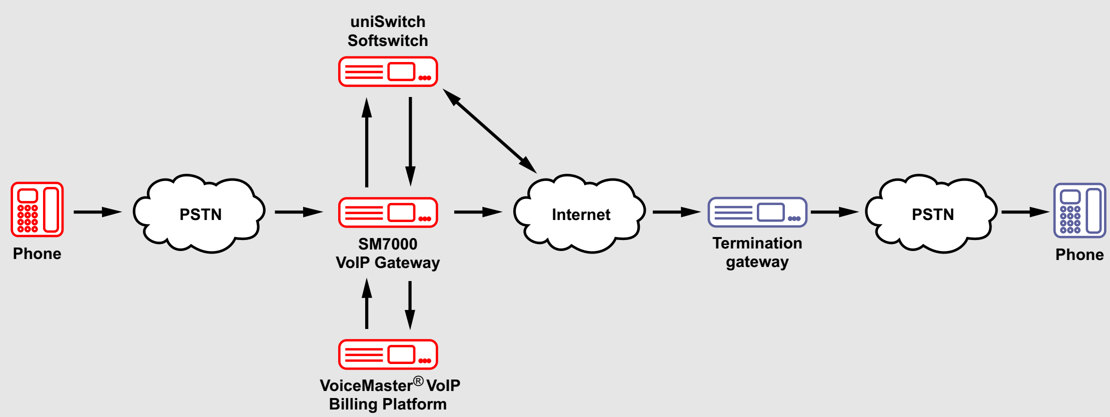
- Deployable in SIP, H.323 and MGCP VoIP Networks
- Registration with Multiple Gatekeepers
- Support for Multiple RADIUS Servers
- Multilingual and Customizable IVR
- Support for IVR over IP
- Multiple Voice Codecs
- Route Fail-over Support
- SIP/H.323 Protocol Conversion
- Codec Translation
- Callback Support

Product Overview

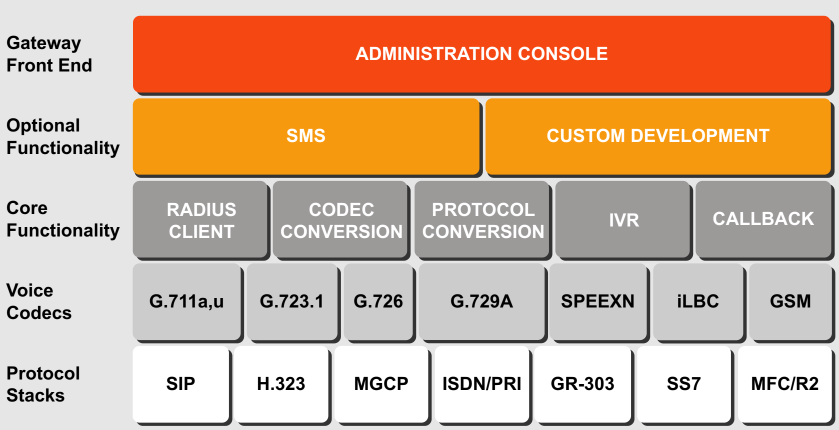
SM7000 is a VoIP gateway that offers universal IP-PSTN switching, carrier grade reliability and high scalability. It supports the popular SIP and H.323 VoIP protocols, and enables service providers to quickly introduce revenue-generating VoIP services, such as calling cards, callback, VoIP termination and others. Compared to other VoIP gateways, SM7000 offers more advanced features, such as callback and IVR over IP, and more compelling return on investment for service providers.

SM7000 VOIP GATEWAY

NETWORK DIAGRAM WITH SM7000



SM7000 PRODUCT ARCHITECTURE



SM7000
VoIP Gateway

Deployable in SIP, H.323 and MGCP VoIP Networks

SM7000 VoIP Gateway supports all major VoIP protocols, including H.323, SIP and MGCP and easily integrates into modern VoIP networks.The gateway also supports multiple PSTN protocols such as SS7, ISDN/PRI, CAS, GR-303, and MFC/R2 to ensure seamless connectivity with virtually any PSTN/SS7 network worldwide.

Registration with Multiple Gatekeepers

SM7000 VoIP Gateway can register with multiple third-party gatekeepers for flawless call routing. This unique feature allows easy product integration with all major providers of call termination services. Gatekeepers can be specified separately for inbound and outbound routing purposes.

Support for Multiple RADIUS Servers

SM7000 VoIP Gateway can operate in complex billing environments with multiple RADIUS servers. For authentication and authorization purposes, the gateway can work with a single RADIUS server; for accounting, however, SM7000 can communicate with multiple RADIUS servers simultaneously

Multilingual and Customizable IVR

SM7000 VoIP Gateway offers enhanced IVR functionality with support for multiple languages and custom prompts. Such functionality enables providers to offer high level of service personalization by configuring SM7000 VoIP Gateway to interact with each subscriber at his/her own language.

Support for IVR over IP

SM7000 VoIP Gateway offers a unique IVR over IP functionality which enables it to encode and transport IVR messages over IP channels to gateways which don't natively support IVR. Such product feature allows service providers to add IVR functionality to their existing VoIP infrastructure with low investment.

Multiple Voice Codecs

SM7000 VoIP Gateway supports multiple voice codecs, including G711, G723.1, G726, G729A, iLBC, SPEEXN, and GSM. All codecs can operate simultaneously on different gateway ports, thus ensuring interoperability with remote gateways supporting otherwise incompatible voice codecs.

Route Fail-over Support

SM7000 VoIP Gateway offers a mechanism to ensure high network availability. The gateway can be configured to periodically conduct L3, L4, and L7 remote service checks and re-route (fail-over) calls to alternative remote gateways if current terminals become unavailable.

SIP/H.323 Protocol Conversion

SM7000 VoIP Gateway ensures maximum interoperability with third party VoIP equipment through its protocol conversion capabilities. The gateway can translate signaling messages from SIP to H.323 and vice versa and thus bridge calls between VoIP equipment using incompatible protocols.

Codec Translation

SM7000 VoIP Gateway offers versatile bridging solution between VoIP terminals supporting incompatible voice codecs. The product can receive voice traffic from the origination gateway encoded in a particular format, decode that traffic and re-encode it in a format supported by the termination gateway

Callback Support

SM7000 VoIP Gateway can be implemented in multiple callback scenarios requiring alternative callback initiation methods. The VoIP gateway can authenticate a subscriber by his/her PIN number send via web, SMS or emailAlternatively, it can initiate a callback based on the subscriber's caller ID (ANI) or DNIS.

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