

SM7000

VoIP Gateway

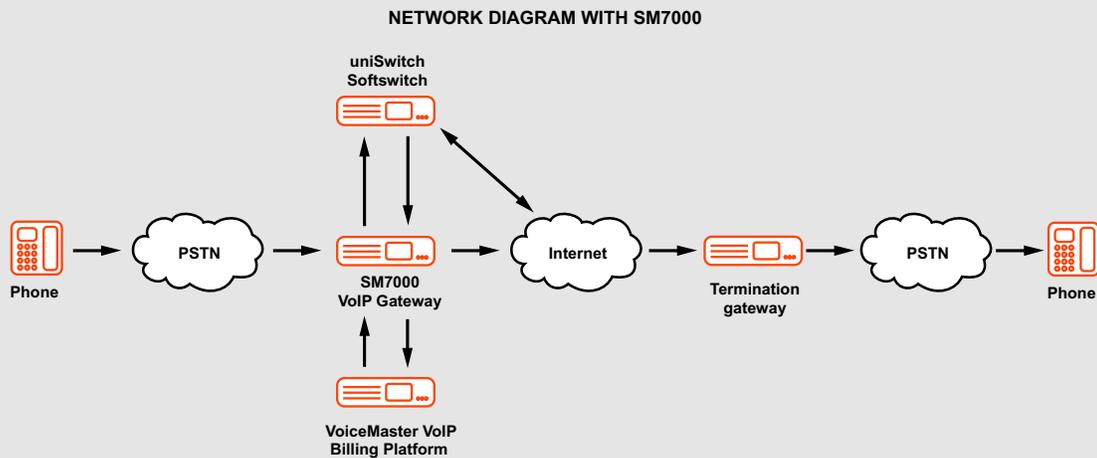
KEY FEATURES

- Deployable in SIP, H.323 and MGCP VoIP Networks
- Registration in 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
- SMS Functionality

Product Overview

SM7000 VoIP Gateway provides both emerging and established telecoms and service providers with an affordable and flexible solution for starting a new VoIP business and/or rolling out new revenue-generating communication services. The product features universal IP-PSTN switching, carrier grade reliability and remote feature upgradeability. Unlike competing products that do switching only, SM7000 VoIP Gateway offers both switching and value-added services, such as callback and SMS. Such product flexibility is valued by service providers as it allows them to start with a basic offering and add new services cost-effectively as they expand their customer base.

SM7000 VOIP GATEWAY



SM7000 PRODUCT ARCHITECTURE

Gateway Front End	ADMINISTRATION CONSOLE						
Optional Functionality	Callback		SMS		Custom Development		
Core Functionality	IVR		RADIUS (client)		CODEC CONVERSION		PROTOCOL CONVERSION
Voice Codecs	G.711a,u	G.723.1	G.726	G.729A	SPEEXN	iLBC	GSM
Protocol Stacks	SIP	H.323	MGCP	ISDN/PRI	GR-303	SS7	MFC/R2

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 in 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 email. Alternatively, it can initiate a callback based on the subscriber's caller ID (ANI) or DNIS.

SMS Functionality

SM7000 can be remotely upgraded with SMS functionality which will allow it to be deployed in a variety of SMS content distribution solutions. In such solutions, SM7000 will receive SMS messages from users, request service authorization from content management servers, and deliver ring-tones, pictures, logos, and text messages to mobile users via SMS.

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