



InterWeave Smart Solutions

SMART TELEPHONY GATEWAY

InterWeave SMART Solutions deliver powerful yet easy-to-use configurable integration Solutions, allowing seamless integration of data from CRM, Call Centers, etc. with your VOIP Telephony devices

InterWeave Smart Telephony Gateway

The **InterWeave Smart Telephony Gateway (ISTG)** solution supports real-time integration with full featured telephony systems running on an IP network. Various IP telephony devices including phones and gateways communicate seamlessly over IP infrastructures integrating data, voice and video with your CRM, Call Centers and other applications that support this integration type. The solution rewards users with greatly reduced cost of ownership and new IP-based feature capabilities using the existing features; micro browsers, etc. that now display accessed data in real-time and in context. ISTG provides the ability to access specific customer records and information in context with the call and how the information has been configured in a "Heads Up" display on your telephony device.

InterWeave Smart Telephony Gateway

ISTG is the interface between your CRM, Call Centers and other applications with your enterprise PBX and a wireless LAN networks that support digital and/or analog line interfaces. ISTG provides data with voice encoding and packetization required to transmit telephone data over an IP network in real-time, working with the necessary control signaling to the host PBX allowing users to access all the features and capabilities of the PBX. ISTG is designed to integrate with the existing enterprise PBX, seamlessly integrating with all of the provisioning, maintenance, and administration functions of the host PBX. The **InterWeave Professional VOIP Telephony Gateway** supports 6 users with the **InterWeave Enterprise** supporting up to 800 Telephone users.

Features

ISTG is designed for the enterprise networks of today and tomorrow. For traditional PBX systems, ISTG provides access to data over traditional LAN's and wireless LAN's without sacrificing digital telephone features and capabilities. To ensure excellent data quality, ISTG has developed features that work with voice prioritization mechanisms for 802.11 access points and Wireless Telephones. ISTG is standards-compatible, simple to implement, and provides real time customer data for viewing.

The **InterWeave Smart Telephony Gateway** offers many features and options that can be tailored to specific customers IP environments. To learn more about how ISTG can support your particular business, please contact us at 203 274 5226, email sales@interweave.biz or go to www.interweave.biz.

Featured Benefits of the Smart Telephony Gateway include:

- Digital data integration with legacy telephone systems
- Full feature access to servers supporting tradition LAN's and Wireless LAN devices
- Data display features of CRM customer and opportunity data in context with Caller ID and message waiting indication
- Scalable Solution architecture starts at supporting 6 users
- Future-proof solution compatible with IP telephony applications

Smart Telephony Gateway Pricing

The **InterWeave Smart Telephony Gateways** are priced starting at \$1400 annually, with bundled pricing available when you include other **Smart Solutions**.

- **Professional VOIP**
 - Starts at \$1,400
- **Premier VOIP**
 - Starts at \$2,400
- **Small Business VOIP**
 - Starts at \$3,400
- **Enterprise VOIP**
 - Starts at \$5,670

Smart Telephony Gateway Detail (next page).



Integration Solutions made Easy

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InterWeave Smart Telephony Gateway (ISTG) Data Flow

The telephony data flow process may be difficult to understand at first. The steps and diagram below illustrates the flow of telephony information from a customers CRM/Call Center application to their Salesforce and back via VOIP PBX.

Step 1: - The first step is the user register. The users must register themselves to be found by other users. In this case, the terminals send a REGISTER request, where the fields "from" and "to" correspond to the registered user. The Proxy server, who acts as Register, consults if the user can be authenticated and sends an OK message if there is no problem.

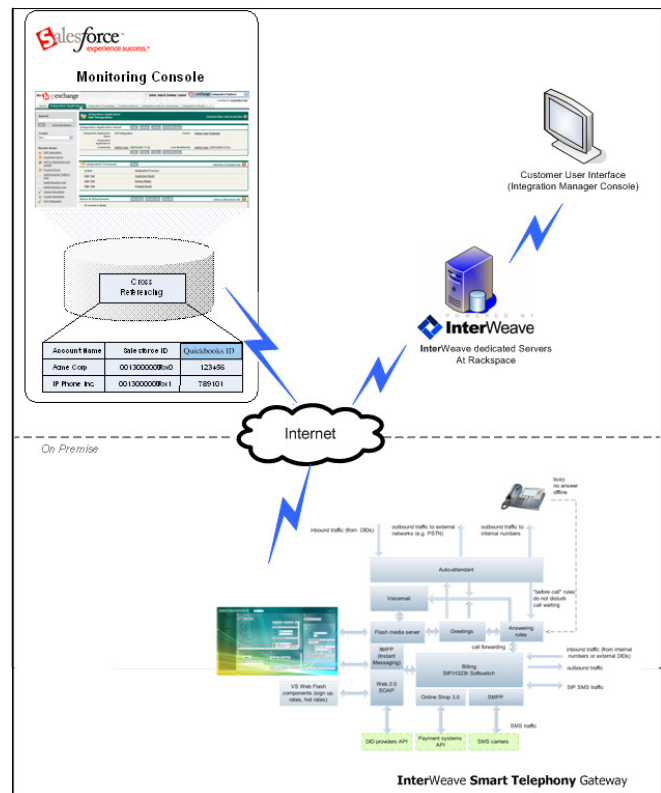
Step 2: -The following transaction corresponds to a session establishment. This session consists of an INVITE request of the user to the proxy. Immediately, the proxy sends a TRYING 100 to stop the broadcastings and reroute the request to the B user. The B user sends a Ringing 180 when the telephone begins to ring and it is also reroute by the proxy to the A user. Finally, the OK 200 message corresponds to the accept process (the user B response the call).

Step 3: - At this moment the call is established, and the RTP transport protocol starts with the parameters (ports, addresses, codec's, etc.) of the SDP protocol.

Step 4: - The Micro Browser request is sent to InterWeave to access Salesforce.com and return customer account detail data (as specified in configuration) which populates the Micro Browser in the device window.

Step 5: -The last transaction corresponds to a session end. This is carried out with an only BYE request to the Proxy, and later reroute to the B user. This user replies with an OK 200 message to confirm that the final message has been received correctly.

Step 6: - Call transcript may be accessed for future reference to call dialogue.



Multiple VOIP Protocols Supported

Megaco H.248	Gateway Control Protocol
MGCP	Media Gateway Control Protocol
MIME RVP over IP	Remote Voice Protocol Over IP Specification
SAPv2	Announcement Protocol
SDP	Session Description Protocol
SGCP	Simple Gateway Control Protocol
SIP	Session Initiation Protocol
Skinny	Skinny Client Control Protocol, (SCCP)