SIP Trunking Workshop for Service Providers

February 1, 2011

The Ingate SIP Trunking and Unified Communications Summit

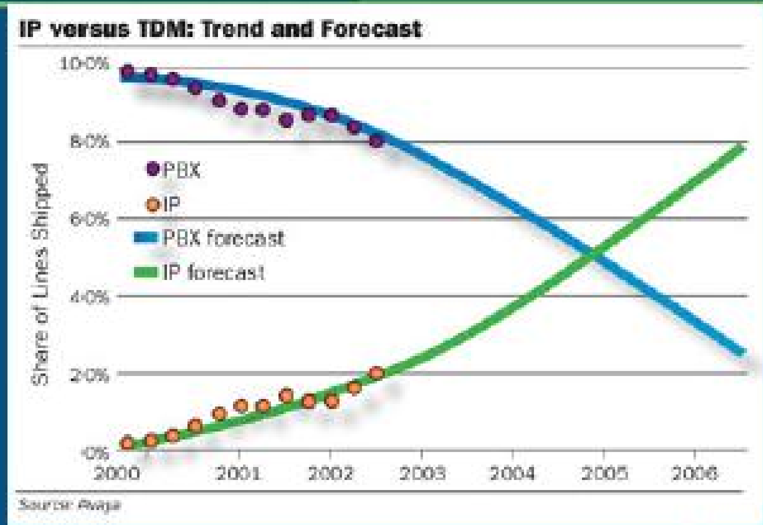
Welcome

Why are we Here?

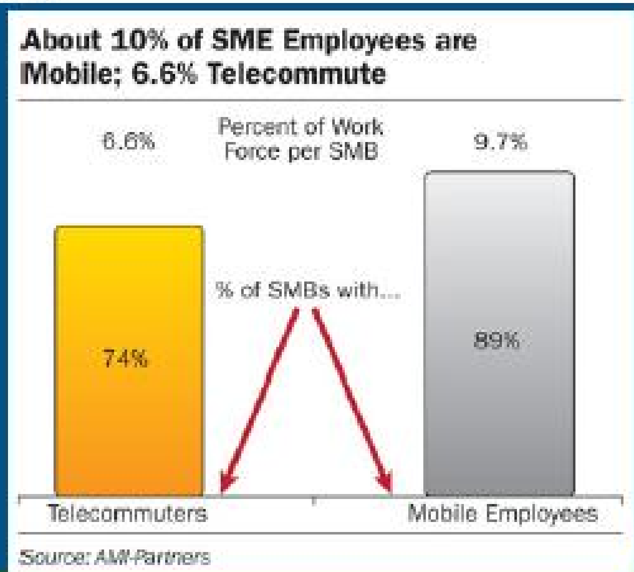
- Education
- Knowledge
- Sharing
- Solutions
- Networking

Economic Reality – an all IP world

- Over 70% of all PBX's sold are now IP enabled, typically SIP based .
- By 2010 50% of the installed base of Enterprise PBX systems will be VoIP.



Telecommuters use VoIP to connect with the office.

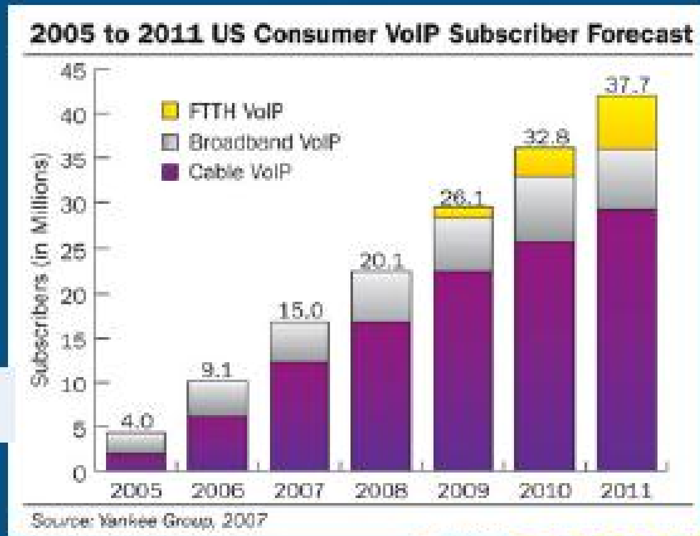


SIPconnect

Economic Reality of Telecom

- Fixed Rate services are dominating telecommunications.
  - Triple Play from Cable Operators –
- All you can eat Fixed rate mobility services
  - Buckets of Mobile Minutes
  - $99.00 voice - text – web
- Variable Costs for Operators have become unacceptable.
  - SS7 dips, for instance

Sell your used Class 5 Switch on EBAY !!!



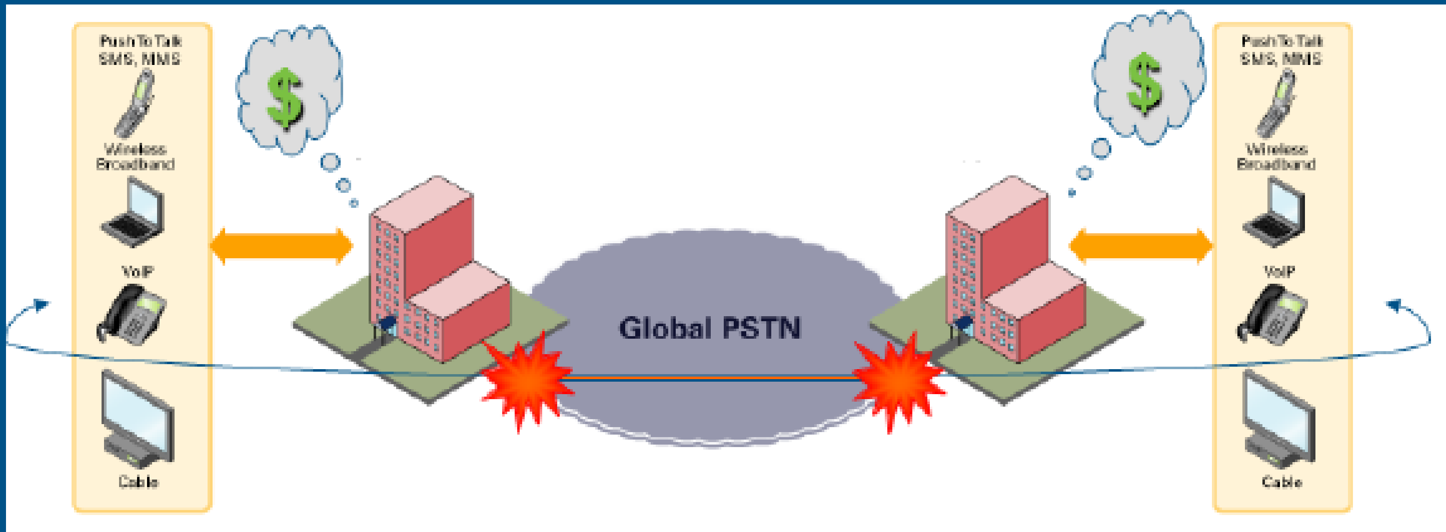
SIP connect

SIP FORUM

The Evolution of Enterprise VoIP

- First : Replace the RJ-11
  - Immediate gains in CAPEX as single wiring harness simplifies campus management.
  - Greenfield ROI – NO Brainer
- Second : Replace the TIE Lines
  - Integrate Enterprise wide Dial Plan Management into single IP Network. Immediate OPEX gains.
- Third : Replace the PRI (Today) SIPconnect
  - All IP E2E
- Fourth : Peer with Business Partners
  - The 40-40-20 rule
- Fifth : Seamless Campus/Mobility Integration
  - Its not fixed Mobile Convergence its Substitution

The PSTN PRI's are the Bottle Neck to new Enterprise Communications services



- The PSTN is used as the inter-VOIP "default" network
  - Service is degraded as it must transverse multiple networks
- Every VOIP network is an Island (apologies to John Donne!)
- PSTN Primary Rate Interfaces are the last bottleneck.

SIPconnect

SIP FORUM

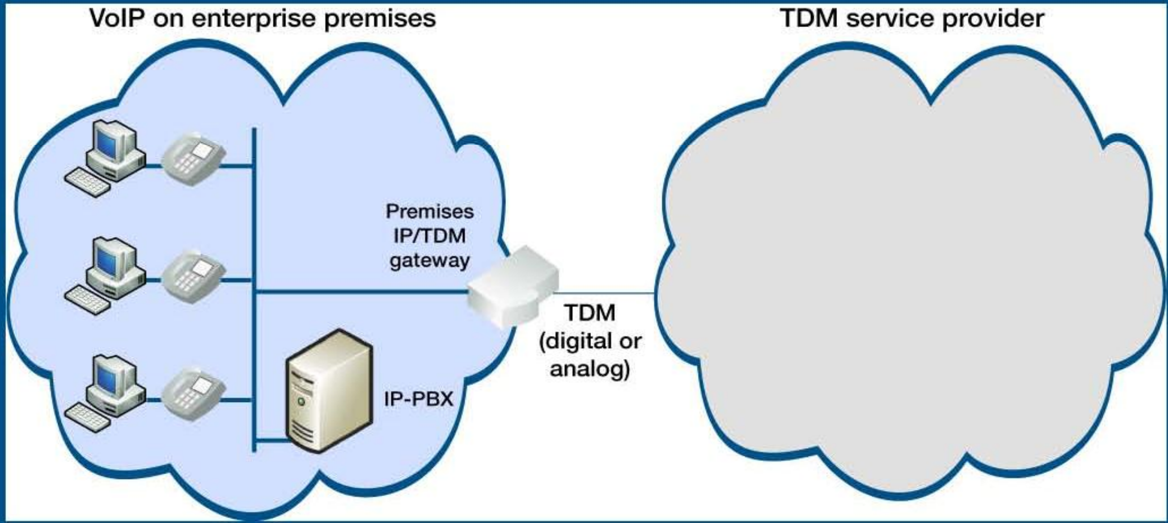
Cost Savings and New Features for Business Customers

- Eliminates TDM gateways and increases efficiency of local access facilities
- Provides DID capabilities w/o requiring the recurring expense of analog lines or expensive digital circuits
- Improves voice quality by removing gateway latency and includes the attentive management of QoS, echo cancellation as well as fax and modem support
- Creates the right foundation for personalized applications and rich media services between customers and service providers as well as between customers and other IP-connected PBXs

Realizing The Promise of IP Communications

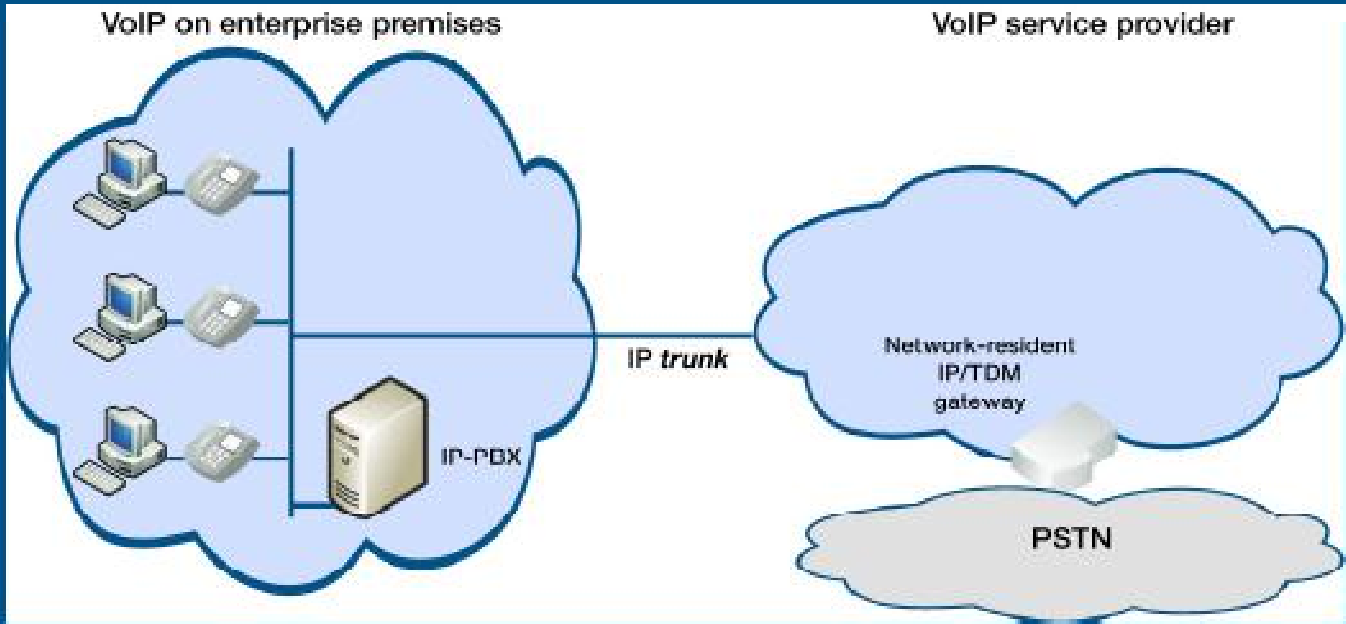
- Problem: IP-PBXs have successfully cut costs and delivered new features to customers, BUT…TDM Routing of VoIP Traffic is a Limited Approach to Achieving Next Generation Telephony
- Opportunity: Preserving and Extending Next-Generation IP Communications Capabilities Beyond the Enterprise
- Solution: Direct IP Peering, or Creating a Seamless, End-to-End Connection between SIP-enabled IP-PBXs and SIP-enabled VoIP Service Provider Networks

The Old Way



- Enterprises must use a gateway to connect IP PBXs to PSTN
  - Increases investment required to move to VoIP, reduces ROI
- Limits ability to leverage VoIP's advantages
  - Adapting to the PSTN means least common denominator functionality
  - Enterprises cannot fully leverage low-cost VoIP termination providers

The New Way



- Connecting IP PBXs directly to VoIP service providers provides significant advantages
  - More features, less cost
- But, how to do it?

SIPNOC 2011

SIPFORUM

SIPNOC – SIP Network Operators Conference

New two-day educational conference for service providers, organized by the SIP Forum, to focus on "making SIP work in the network" and address the key operational issues facing the deployment of SIP in today's wireline and wireless networks

SIPNOC Topics

- Application development
- Testing
- SIP Trunking
- FoIP
- User-Agent Configuration
- Emergency Services
- Policy Servers
- Security
- Operational Issues
- Call Routing and Peering
- Troubleshooting and Monitoring
- SIP Interconnection (SIP-to-SS7 signaling gateways)
- HD-Voice/Video Deployment Challenges

- April 25-27, 2011 at Hyatt Dulles Hotel, Herndon, VA
- Developed for the technical and operational staff of service providers, including engineers.
- Features special keynote from Dr. Douglas Sicker, Chief Technologist, FCC
- Dr. Eric Burger serving as Program Chair
- Early-Bird Discount and Special 50% Discount for SIP Forum Full Members (min 2 registrants from Full Member company)

SIPNOC 2011

- Event Website: **http://www.sipnoc.org**
- Register at: **http://www.regonline.com/sipnoc_2011**
- Hotel Reservations: **https://resweb.passkey.com/go/SIPNOC2011**

Benefits of SIP Trunking

- Monthly cost savings
- Single network for all communications
- Lower cost of Moves, Adds and Changes
- Disaster Recovery / Business Continuity
- User provisioning

But Let's consider this as well:

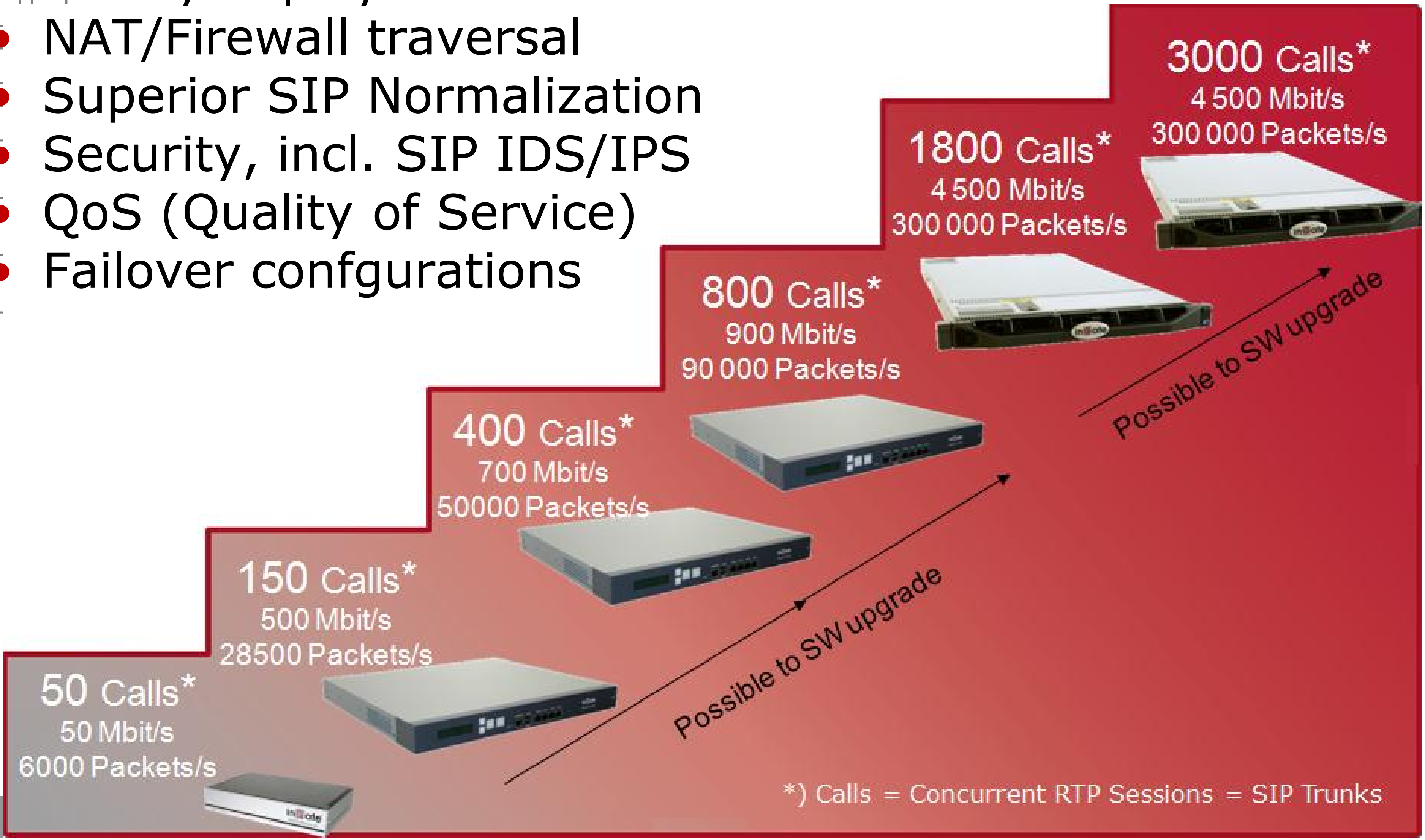
- Steps of going beyond POTS replacement – Unified Communication
  - Mobility – Remote workers
  - Multimedia - Video, better Voice, IM, Presence, Real Time Text RFC 4103, etc.
  - Real SIP address – like email address
  - WiFi mobile phone communication

Intertex & Ingate

- Same parent company
- Intertex for service provider volume deployments
  - Session Border Controllers for SMB and home offices
- Ingate for enterprise and project installations
  - The Ingate Firewall and SIParator® product lines
- Cooperation in management and development
- Co-developed SIP code
- Ingate represents Intertex in the US

The Ingate Family of Enterprise SBCs

- From 5 to 3,000 simultaneous calls
- Widely deployed
- NAT/Firewall traversal
- Superior SIP Normalization
- Security, incl. SIP IDS/IPS
- QoS (Quality of Service)
- Failover confgurations



The Intertex IX78 E-SBC

- ADSL2+ modem with Annex A/B/M (24 Mbps DS, 3 Mbps US), or
- Ethernet WAN (VLAN capable)
- Triple play and various routing configuration possibilities
- Router with any port, any service capability and 4-5 port Ethernet Switch
- Wireless 802.11b/g as Access Point (3 SSID for separate WLANs)
- Business Firewall
- Advanced QoS for voice, IP-TV etc.
- VPN (IPsec with certificate handling)
- TR-069 and proprietary flexible provision system

and in addition to VoIP things like

➢ 2 FXS ports for analog telephones and FAX with T.38 support

➢ FXO port: Real SIP/PSTN gateway + Fallback on WAN loss

there are outstanding features enabling new applications and services

➢ **Unique support for standard SIP phones and soft clients on the LAN and WLAN**

➢ **SIP Trunking of PBXs – unequalled interoperability list**

➢ **SIP Proxy, Registrar and PBX-like functionality**

and more…

Now it is about SIP Trunking Services from a Service Provider perspective

		MODERATOR	ORGANIZATION	SPEAKER
1:00pm	The Case for SIP Trunking	STEVEN JOHNSON	Ingate	STEVEN JOHNSON
1:30pm	Delivering SIP to the Enterprise	JOEL MALOFF		
			Broadvox	CHAD KRANTZ
			Intertex	KARL STAHL
2:30pm	The Value of a Service Provider Demarcation Point	JOEL MALOFF		
			EarthLink Business	SCOTT YELTON
			Intertex	KARL STAHL
3:30pm	Ensuring Interoperability -- The Key to Service Revenue Growth	JOEL MALOFF		
			Bandwidth .com	SEAN RIVERS
			Intertex	KARL STAHL
3:50pm	BREAK			
4:30pm	Addressing Security Issues	JOEL MALOFF		
			Ingate	SCOTT BEER
			Intertex	KARL STAHL
5:30pm	Generating Revenue from HD Video	JOEL MALOFF		
			UCIF	STEFAN KARAPETKOV
			Intertex	KARL STAHL
6:30pm	Wine Reception			

Who is here?

Audience – Show of Hands ☺

- Service Provider or Operator?
  - ITSP?
  - Telco (having their own delivery network)?
- VAR or Retailer?
- User?

- How many are familiar with SIP?
- How many of you are familiar with our products?
  - Ingate?
  - Intertex?

PLEASE FEEL FREE TO ASK QUESTIONS!