SIP Trunking

Overview

- *Introduction*
- *Technology phase shift - TDM to IP*
- *SIP Trunking*
- *SIP Trunking Issues*
- *Solutions*
- *Conclusion/ Questions*

SIP Trunking

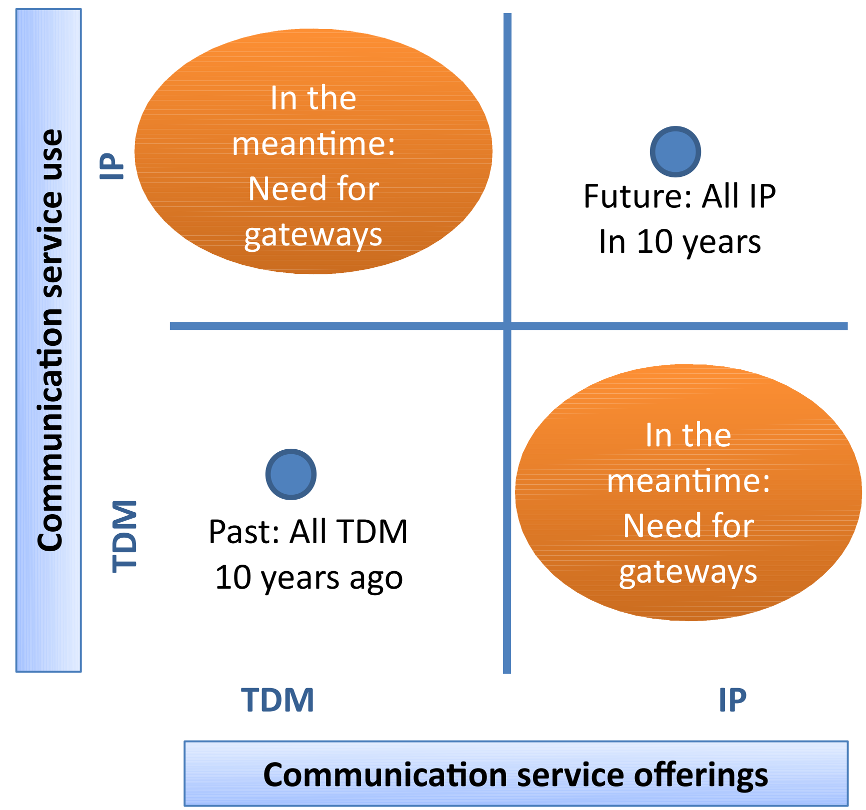
- SIP Trunking is basically routing calls over the IP Network of the provider (ITSP)

- SIP Trunking features typically include
  - Integrated Local and Long-distance calling
  - Unified Communications
  - E911 type Emergency Calling Services

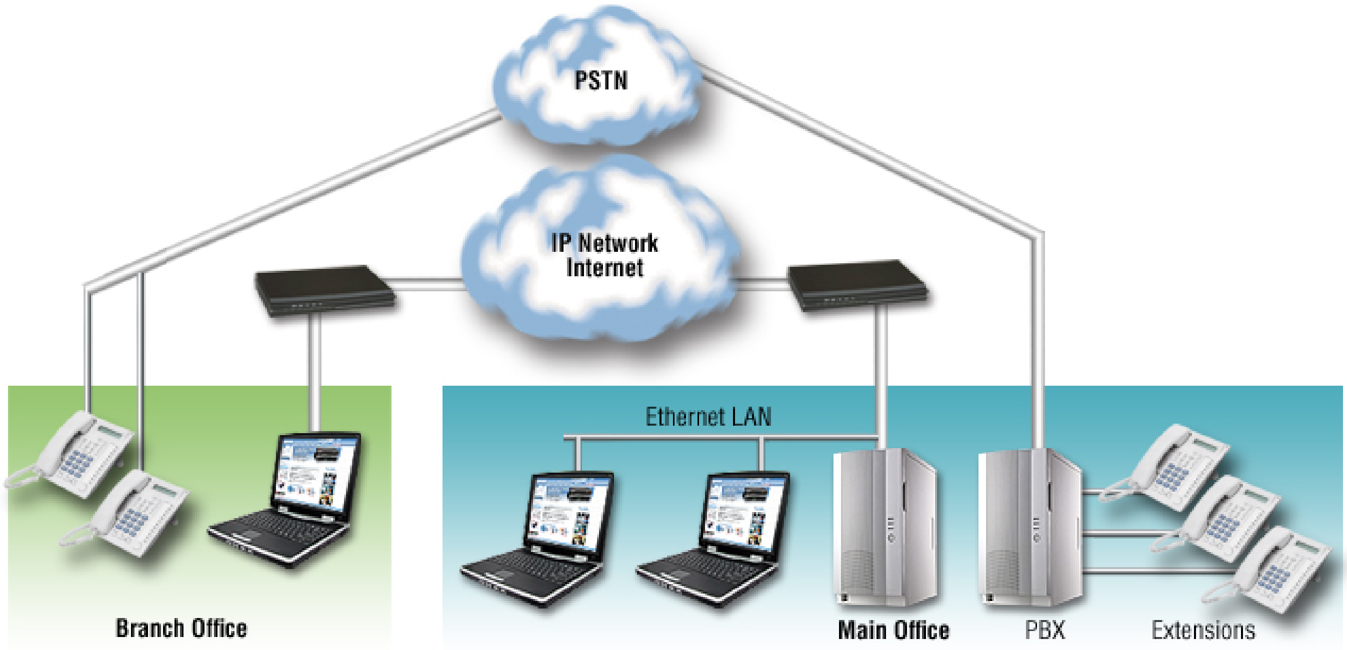
Technology Phase Shift – TDM to IP

Technology Perspective

- Technology shifting from TDM to IP
- Service offerings and usage are changing at different paces
- Resulting transition phase is the current VoIP market

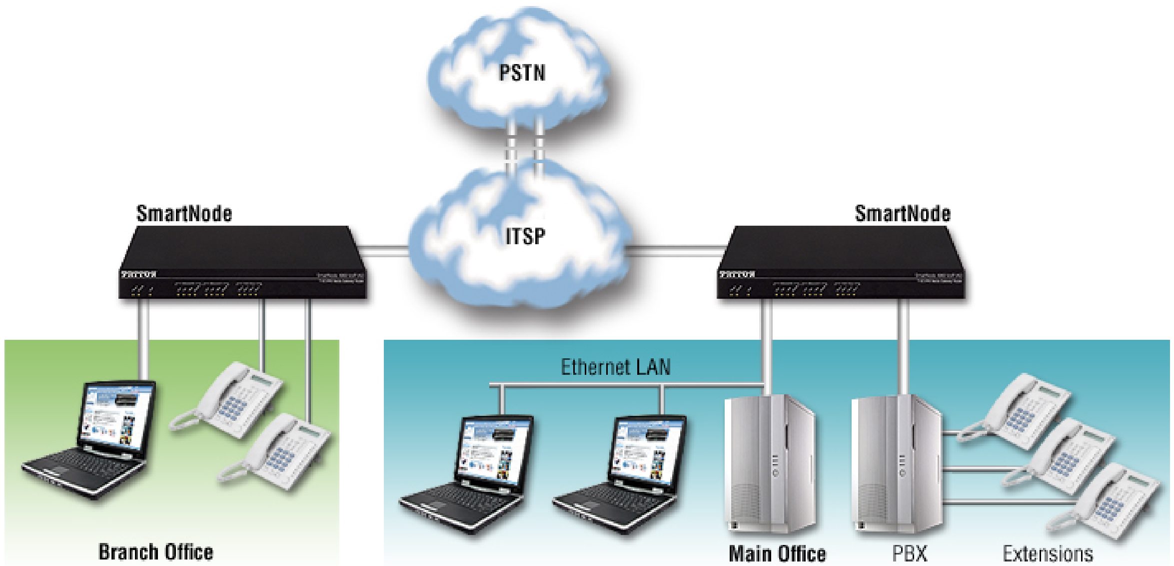


Old PSTN and Data Network



- Three Bills every month for the Business User
  - Local Calling
  - Long Distance Calling – Most Expensive still

SIP Trunking



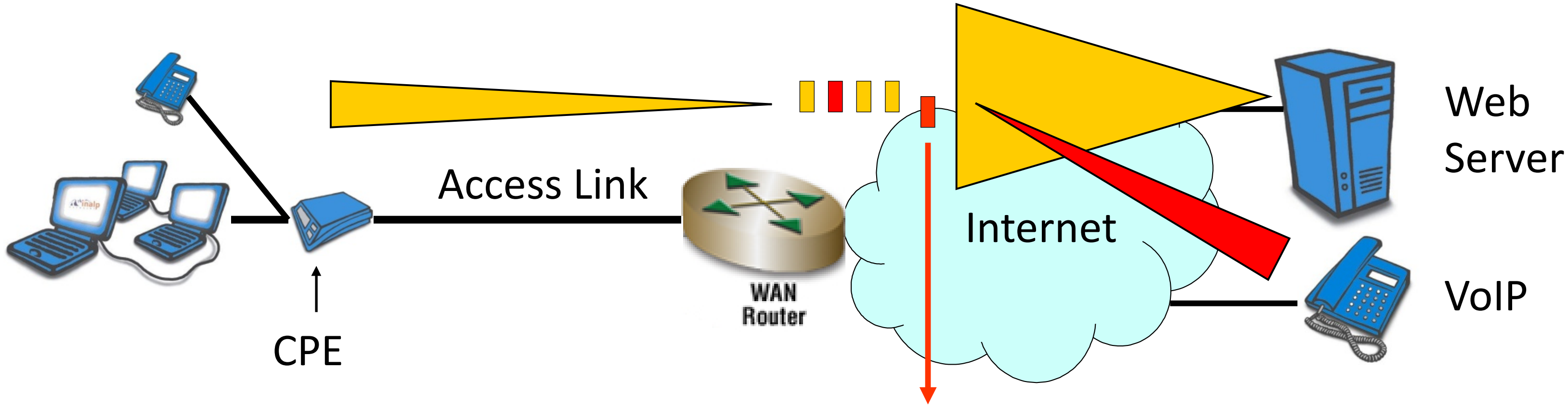
- Single Bill for the Business user
  - Reduced Long distance calling ($$$)
- Enable legacy or IP PBX to use VoIP
- Create a single conduit for voice and data
- Enable value added Enterprise services

Why offer SIP Trunking?

- Infonetics predicts a CAGR of 89%
- Bind Customers through Service Differentiation
- Bind Customers with Bundled Services
- Cut Operation Costs through Converged Service Delivery
- Increase ARPU and Reduce Churn
- So What's stopping you?

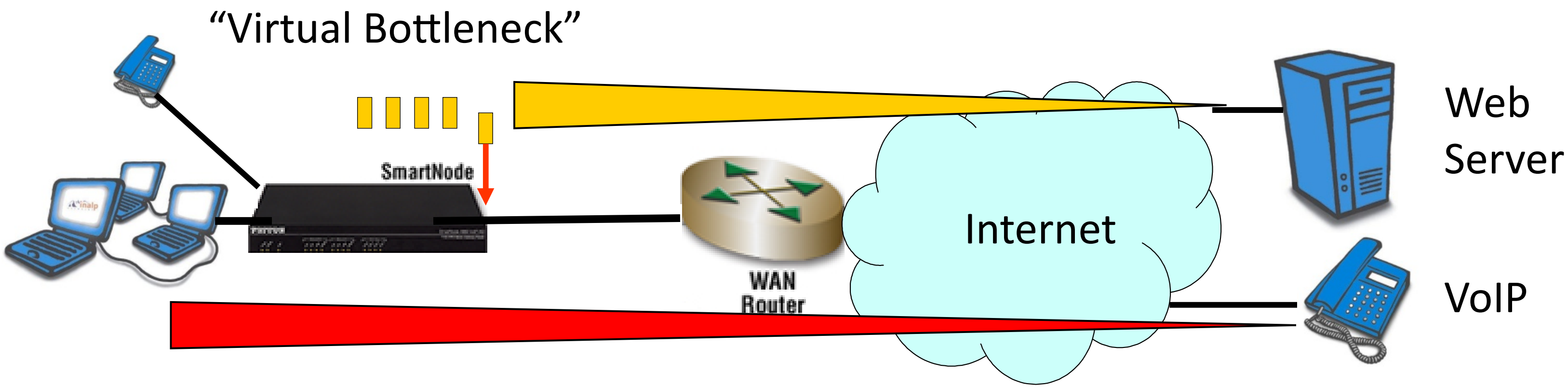
SIP Trunking Issues: General

Issue: Downstream Voice Quality



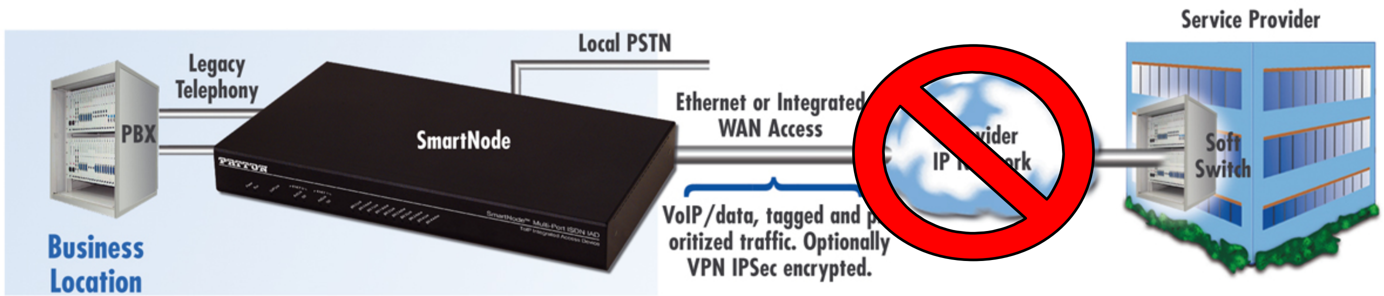
- The Internet only supports best effort packet forwarding
- No differentiation between time critical IP packets such as VoIP and other traffic such as web pages mail etc
- Problem:
  - Downstream is overloaded
  - VoIP and other packets get discarded with same

Solution: SmartNode DownStreamQoS™



- SmartNode DownStreamQoS™
- Introduces a dynamic virtual bottleneck at the customer premises
- Discard non-real time traffic <u>before</u> it starts to block the voice traffic in the edge router

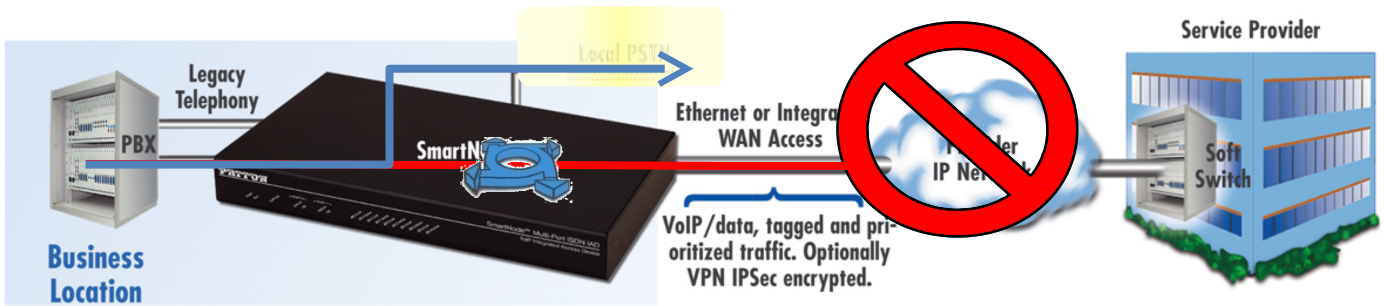
Issue: Survivability



- The Legacy PSTN network has high availability/ survivability features
- Enterprise want survivability in SIP trunking as well
- Enterprises care about this because
  - Business continuity and thus communication continuity is key
  - If the IP network fails, business communication is down

Solution: SmartNode Survivability Features

- SmartNodes pioneer in Telephony Survivability
- Leading Call Routing / Fallback Engine



- Survivable paths for voice

Issue: Network Management

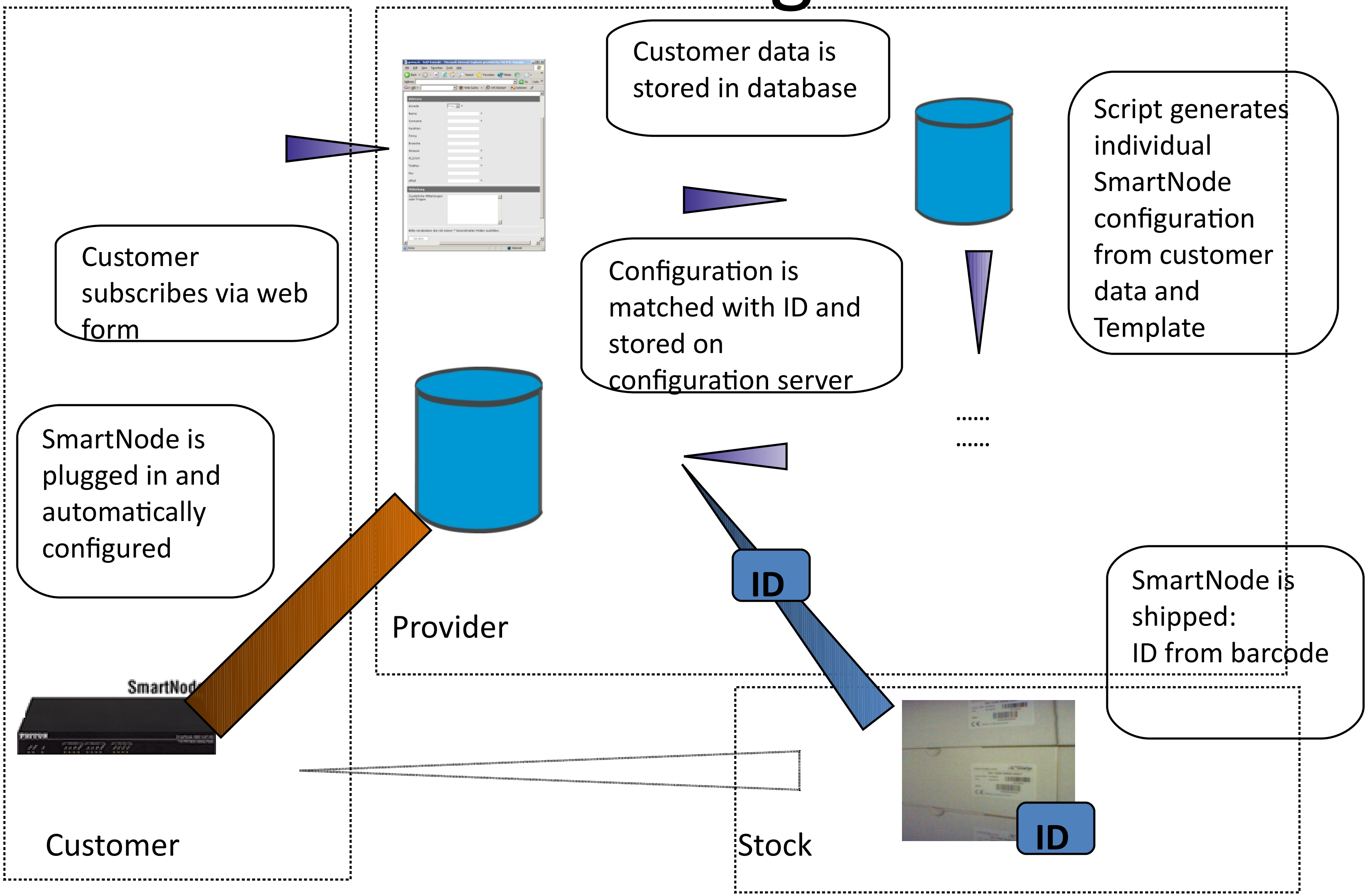
Problem

- Configuring and maintaining a large number of VoIP CPEs is time consuming and expensive
- The VoIP service customer is not trained to do the maintenance of his/her SmartNodes

Solution

- Auto-Provisioning
- No manual configuration effort by the provider or the customer

Solution: SmartNodes Auto-Provisioning



Fax over IP

1) Fax Bypass, G.711

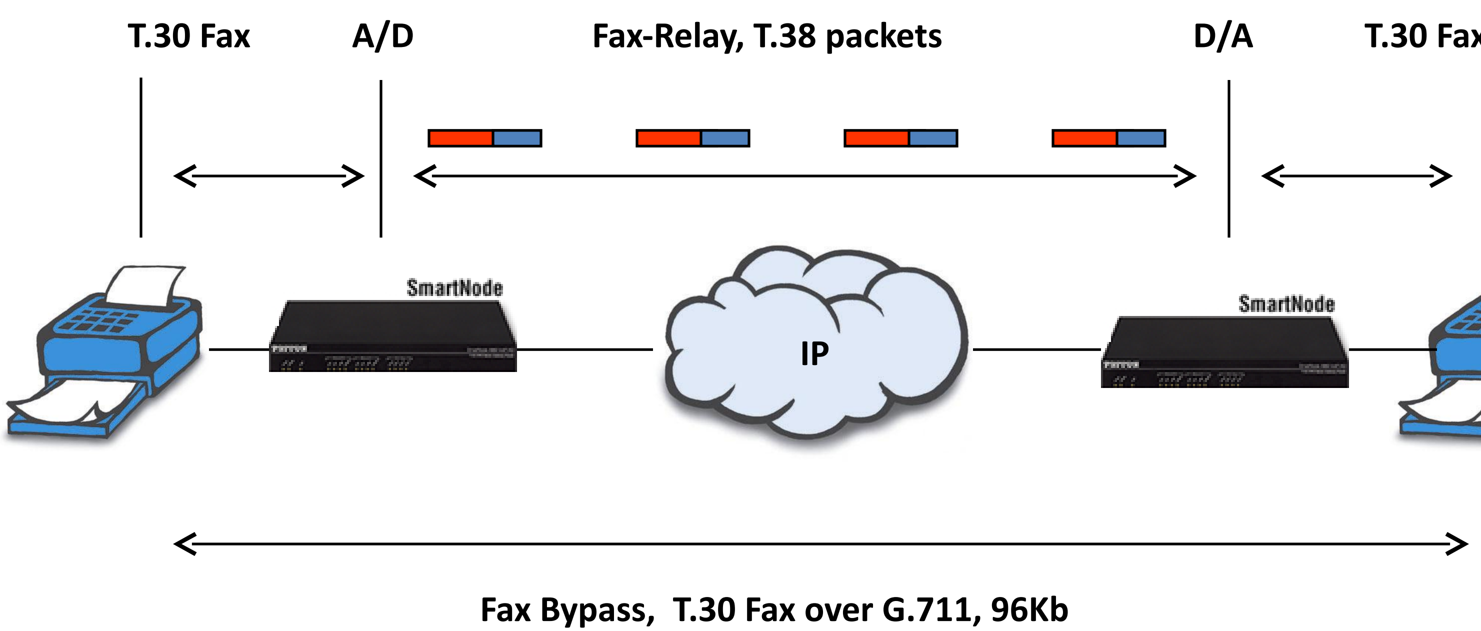
The Fax is carried in a G.711 voice channel just like a regular phone call.

+ Is interoperability with any gateway
- Uses more bandwidth
- Is less reliability

2) Fax Relay, T.38

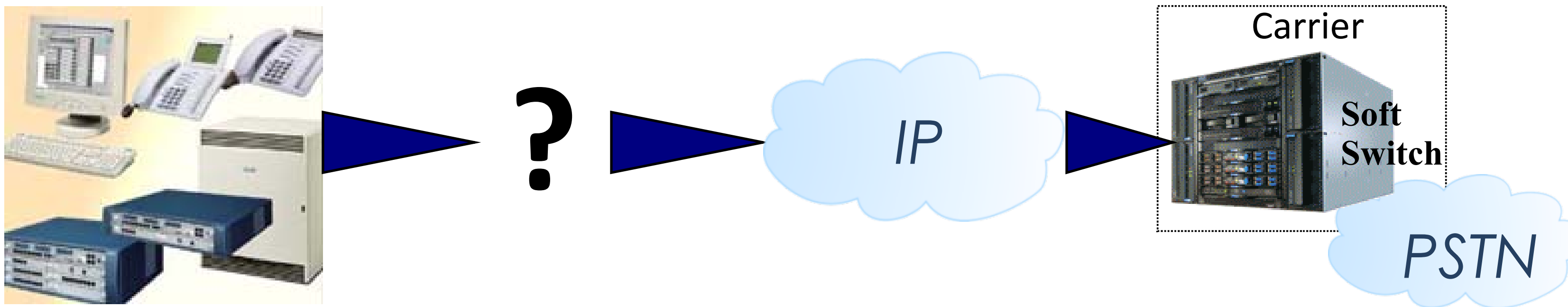
The Fax tones are terminated in the gateway, relayed in packet form and re-modulated at the far end.

+ Uses less bandwidth
+ Is reliability(offers redundancy)



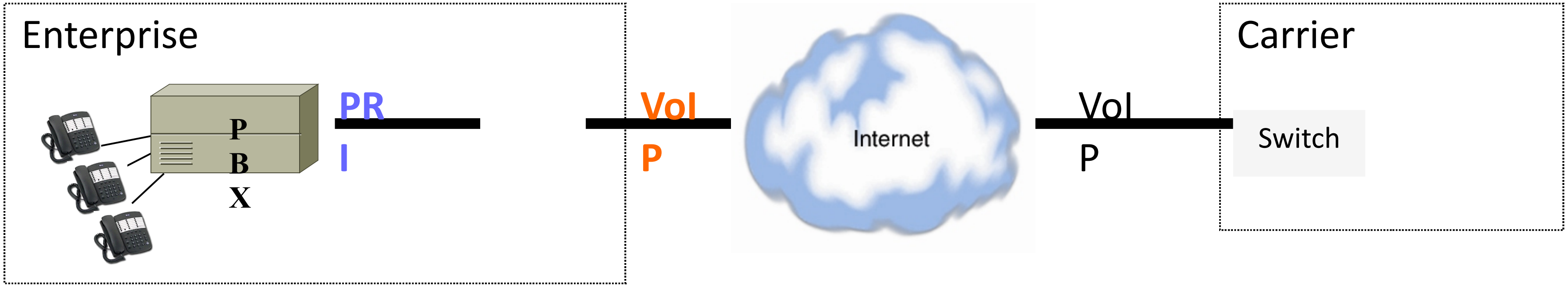
SIP Trunking Issues: Legacy PBX

Legacy PBX Issues



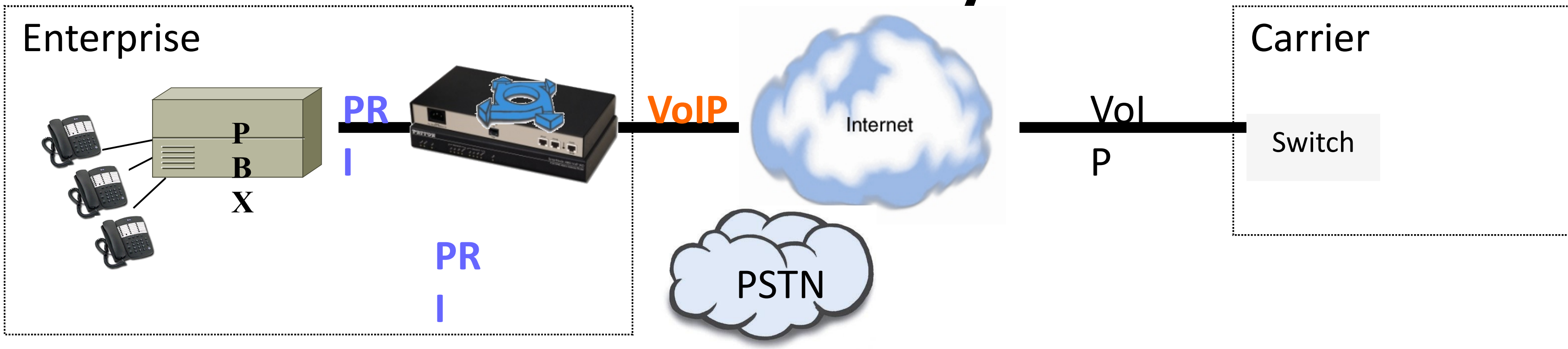
- Legacy PBX is a huge Investment
  - Enterprises would still like to use their old PBX
- Legacy PBX doesn't understand IP
  - SIP/ VoIP are IP based offerings

Legacy PBX Issues



- Needs:
- Integrate different legacy PBX's using a single platform
- Enable SIP Trunking in Legacy PBX
- Phased Migration for the Enterprises
- Solution:

Solution: SmartNode SN4960 Gateway



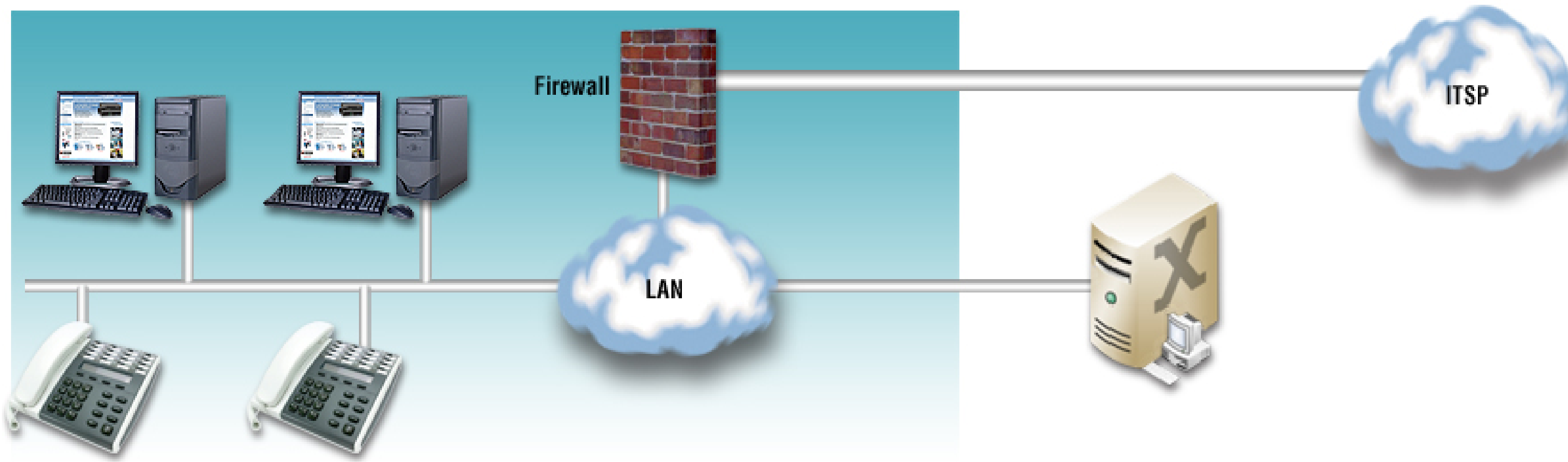
- Use legacy PBX while enjoying benefits of SIP Trunking
- End users don't realize they are using VoIP
- Easy Phased Migration
  - Simple Number Portability using the SN4960 Call Router
  - Run IP and legacy services together during migration
  - Move to IP services completely when desired

SIP Trunking Issues: IP PBX

IP PBX Issues: Interoperability

- SIP is an Open Standard protocol
- Typically interpretation and implementation of SIP is different for different parties involved
  - Many SIP methods/ extensions might be supported by an IP PBX but not by an ITSP and vice versa
  - Enterprises might want to use different codecs for different scenarios (e.g. G.711 for Internal calls and G.729 for External calls)
- Need: A mechanism to ensure interoperability between the parties involved (e.g. ITSP and the IP PBX)

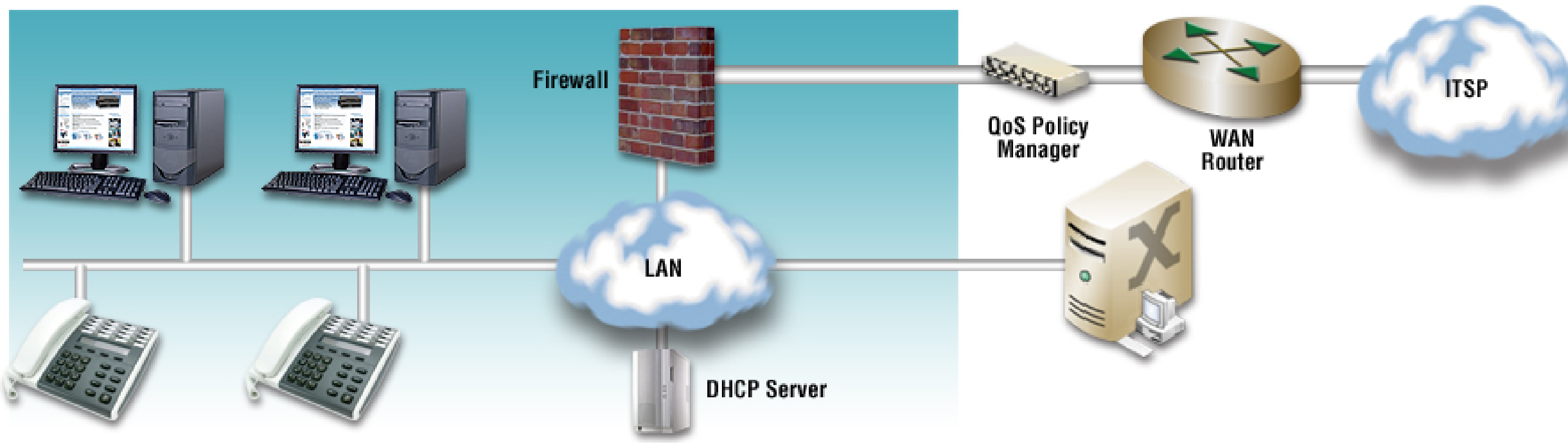
IP PBX issues: Survivability



- Survivability for an Integrated Access Device
- Data Network Design is always Survivable
- In above case, Calls start failing if IP PBX fails
- Calls start failing if connectivity to ITSP fails

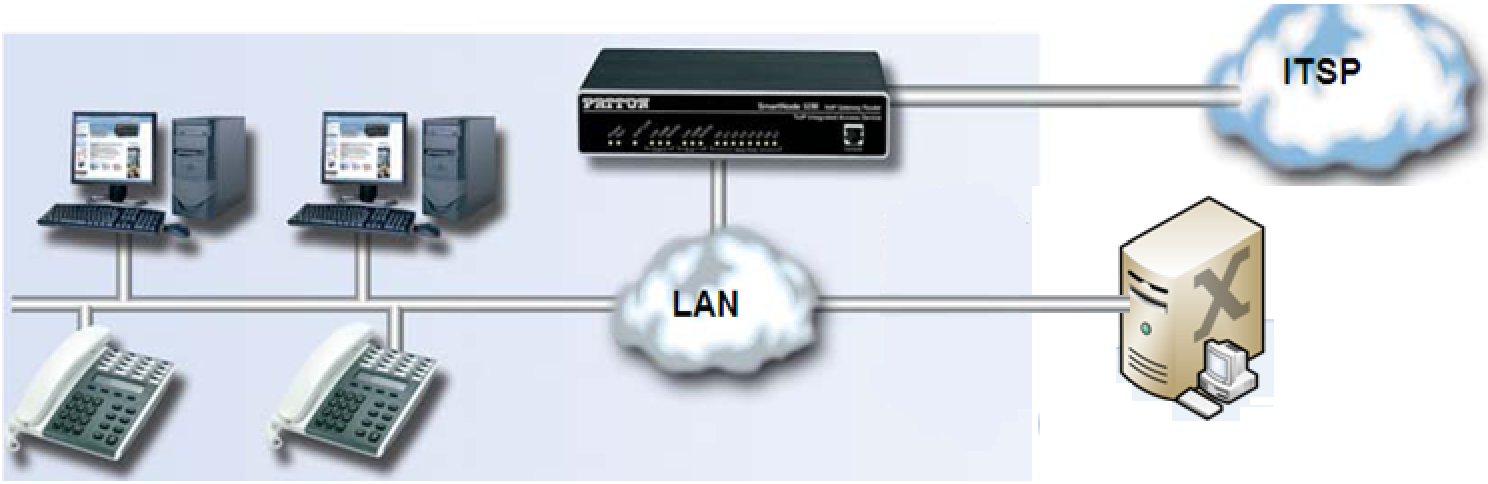
- Need: Survivable telephony based SIP Trunking

Multiple Requirements at the Edge



- Typical Requirements needed to be satisfied
  - QoS
    - Using Data and Voice together mandates an Active QoS mechanism
  - WAN
    - Need for a separate WAN router
    - Efficient utilization of WAN bandwidth for LAN to WAN calls
  - DHCP server

Solution: Session Border Controller SN5200

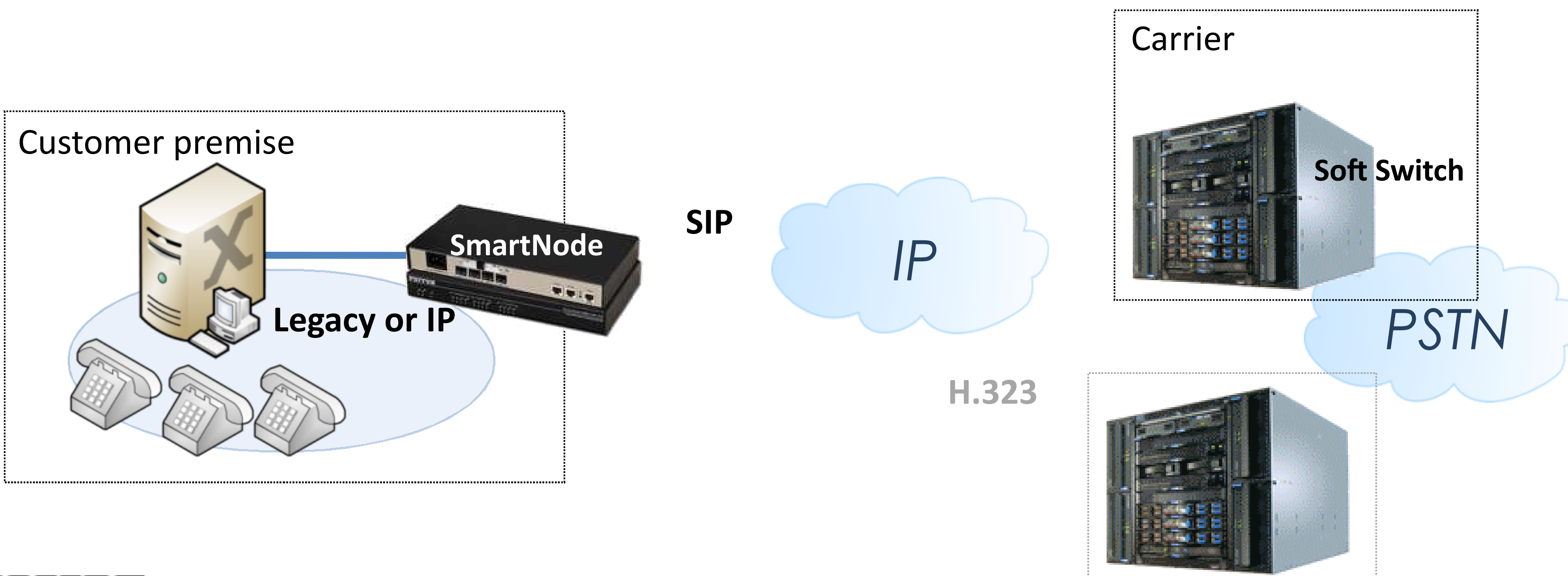


- Enable up to 32 SIP sessions simultaneously
- SIP Session Negotiator for SIP Trunking Services
  - Use appropriate Codec depending on the nature of the call (LAN to LAN or LAN to WAN)
  - Mediate SIP extensions and methods between the ITSP and the IP PBX
- Downstream QoS to handle Voice and Data Traffic
- Secure Enterprise VPN Router Solution
- Integration with your existing Firewall

SmartNode as a Service Provider CPE

Connecting a PBX to a VoIP operator

- ## Scenarios
  - TDM PBX to SIP
  - ISDN PBX to SIP
  - Analog to SIP
  - IP PBX to SIP
- ## Advantages
  - Active QoS traffic shaping
  - Survivability
  - Fax and POS support
  - Special DECT functionality
  - Smooth migration



SmartNode as a Service Provider CPE

Example

- Widely deployed since 2001
- References
  - UPC
  - T-Com
  - O2 Business
  - Avantel
  - Welcome Italia
  - Telekom Austria
  - Telkom South Africa
  - And Many More

- Thank you