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User to Network Interface

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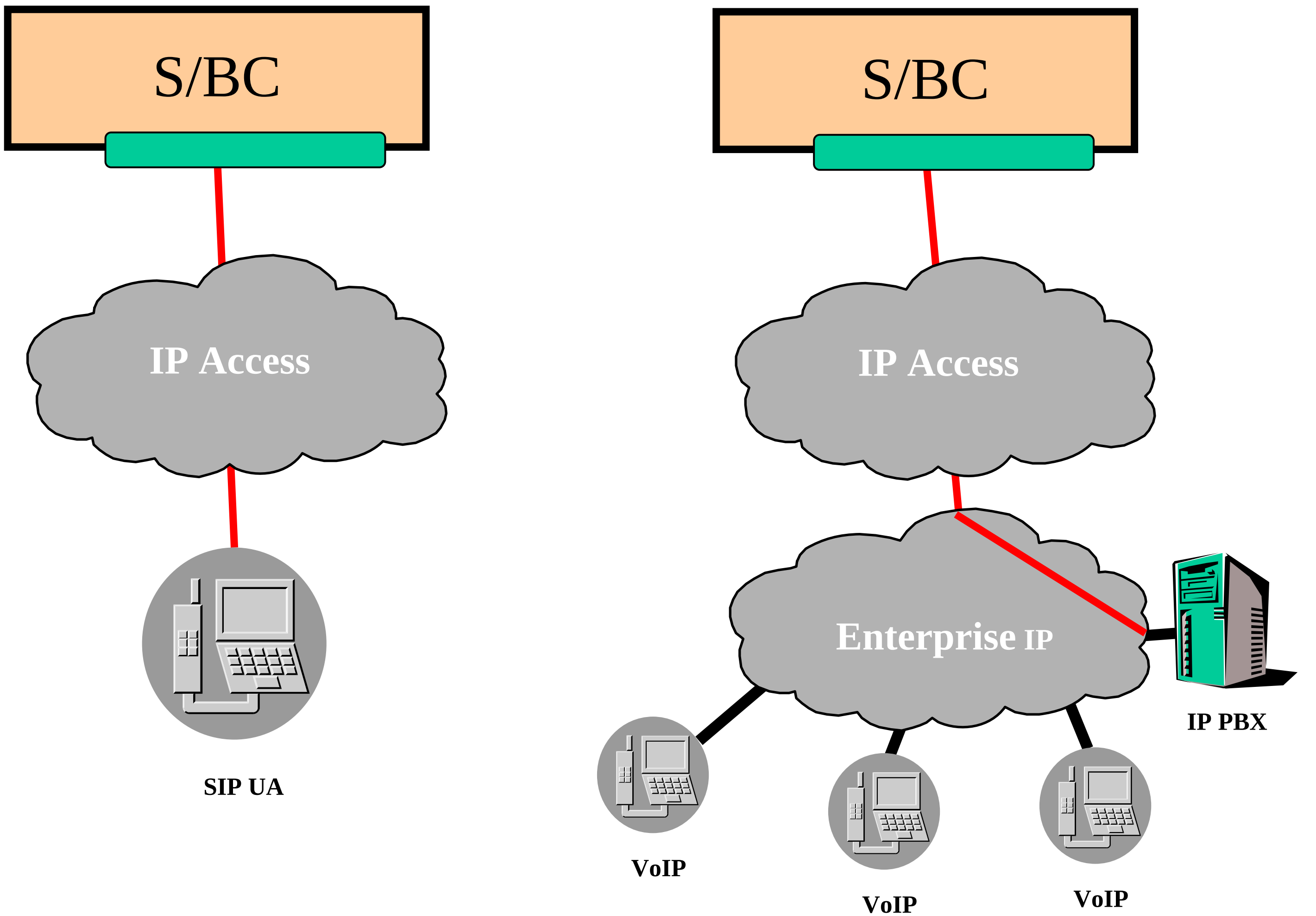
Scope

- Requirements for how Users Connect to Service Providers (UNI)
  - Focus is on the VoIP Layer (Media and Signaling)
  - Independent of whether the VoIP Service Provider is also the IP Provider or whether the IP Provider is a 3rd Party
- Types of User to Network Interfaces
  - Individual Endpoints

    e.g. SIP Phones, Analog Telephone Adapters
  - Aggregate Endpoints

    e.g. IP PBX

UNI Types



S/BC
IP Access
SIP UA
S/BC
IP Access
Enterprise IP
IP PBX
VoIP
VoIP
VoIP

UNI Requirements – Individual Endpoints

- Support Baseline Profile for Interoperability
  - Relevant SIP RFCs
  - Standardized Limited codec set
  - Interact with IP QoS mechanisms
- Support for Usability and Service Provider customizations
  - Device Configuration and Bootstrapping
  - Service Provider Call flows (standardized and/or customized)

UNI Requirements –Aggregate Endpoints

- Support Baseline Profile for Interoperability
  - Relevant SIP RFCs
  - Standardized Limited codec set
  - Interact with IP QoS mechanisms
- Support interoperability with IMS
  - Expose Individual Endpoints
  - Service Provider Call flows (standardized and/or customized)

SIP Requirements - Individual and Aggregate

- UNI needs to support relevant SIP elements that enable a broad variety of Basic and Advanced Voice Services such as:

Call Waiting	Caller ID	Caller Name
3- Way Conferencing	DID/DOD	Locate-me
Billing/Acct Codes	Call Forwarding	Call Hold
Mid-Call Move	Voicemail (UM, MWI)	Call Park
Presence	MS Office Integration	IM

Common Codec Set - Individual and Aggregate

- Objective is that Endpoints can negotiate a common codec and avoid transcoding
- Common codec set should include:
  - Good legacy PSTN support
  - Codec that provides good quality at low bit rates
  - Common codec across Wireline/Wireless
  - High-quality codec that would drive VoIP adoption and advanced application (like enhanced conferencing).
- Good Candidates:
  - Three legacy codecs (G.711, G.726-32, G.729)
  - One good low-bit-rate codec (GSM-AMR NB)
  - One wideband, high-quality codec (GSM-AMR WB)

Usability and Customizations – Individual Endpoints

- Device Configuration and Bootstrapping
  - User should not need to configure Endpoint
  - Endpoint should automatically obtain latest configuration and/or firmware from Service Provider servers (at start-up and periodically)
  - Firmware/Configuration data could include information about custom call flows (e.g. 3-way calling in network or device)
- Models Supported
  - Service Provider supplies Endpoint
  - Customer purchases Endpoint Retail
  - Authentication and Linkage between Endpoint and Customer should be automatic

IMS Support - Aggregate Endpoints

- IMS requires that Endpoints Register to make and receive calls
- IMS supports Implicit Registration that allows a single Registration of a Public/Private Identity to cause the Registration of an Implicit Private Registration Set
- IP PBXs today do not Register
- IP PBXs should Register
  - Could Register the Individual Endpoints
  - Could use Implicit Registration

Key Takeaways

- UNI for Individual Endpoints needs to support
  - SIP Call Flows
  - Standard – small Codec Set
  - Device Configuration and Bootstrapping
- UNI for Aggregate Endpoints needs to support
  - SIP Call Flows
  - Standard – small Codec Set
  - Registration

Thank You