Service requirements

- Two interconnected NGN operators offer end-to-end services
- Quality of Service must be guaranteed in end-to-end fashion through coordination among operators
- Example: carrier-grade VoIP services

VoIP Service requirements

- ETSI TISPAN identifies the emulation/replacement of PSTN/ISDN services as a key issue of the
- With service emulation, a new service is provided through the NGN with identical features of the old service
- Replacement means that some features of the new service may be slightly different

VoIP Service requirements

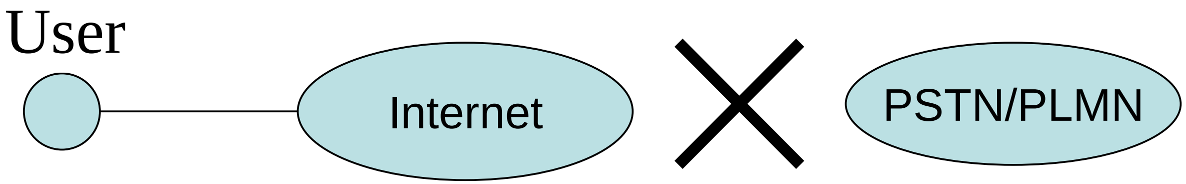
- Basic requirements of the classic PSTN/ISDN service:
  - Numvering plan must be preserved
  - Lawful Interception (LI) must be guaranteed
  - Emergency services must be guaranteed
  - Malicious Call Identification (MCID) service must be guaranteed
  - Anonymous Call Rejection (ACR) service (LI) must be guaranteed
  - Interoperability with the old PSTN/ISDN service must be guaranteed
- Two basic categories of telephone services are devised:
  - **Publicly Available Telephone Service** (PATS);
  - ECS (**Electronic Communication Service**).
- PATS is the service mapping for the classic PSTN/ISDN service and has more tight requirements than ECS

VoIP Service requirements

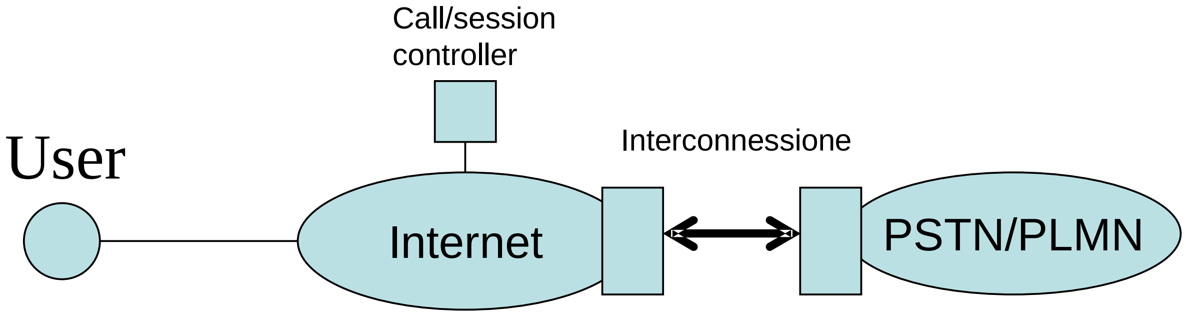
- The emulation of the classic PSTN/ISDN service is usually referred to as ToIP (Telephony over IP), to distinguish it from ECS services, such as VoIN (Voice over Internet)
- In the VoIN service network operatori usually do not control the service and do not guarantee qos
- Typical VoIN services are p2p telephony such as Skype, among others
- The VoIN In&Out service allows users to interconnect through external networks such as PSTN/PLMN

VoIP Service requirements

a) VoIN Peer-to-Peer



b) VoIN In&Out

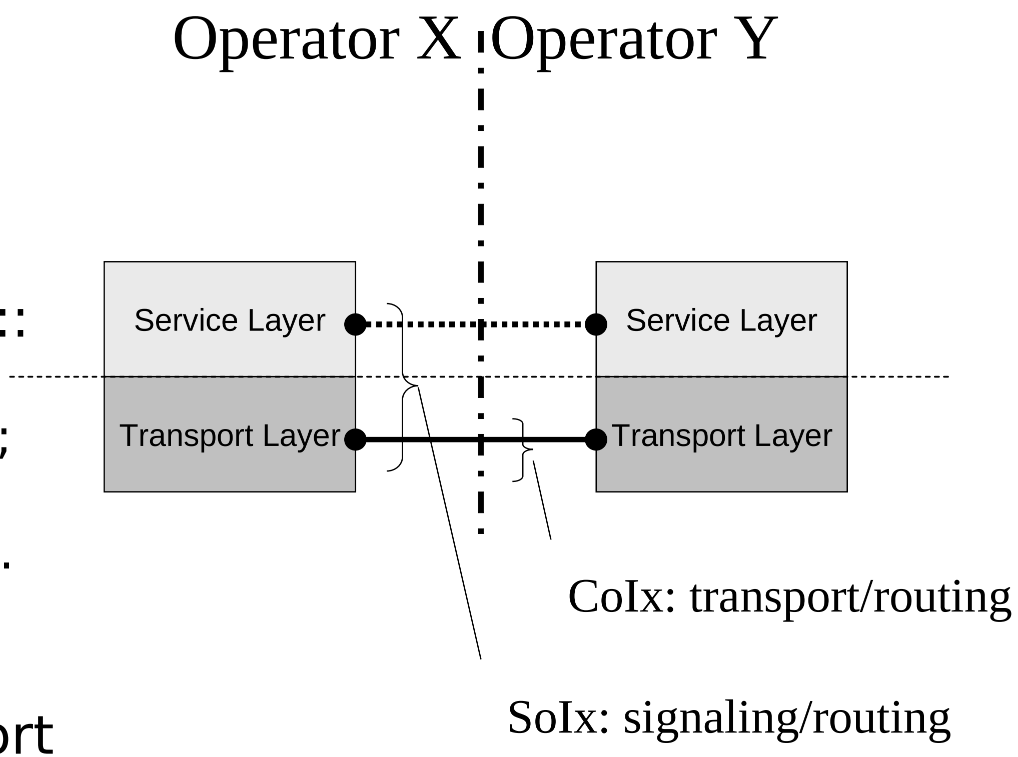


VoIP Service requirements

	Calls to PSTN & PLMN	Additional services (emergency calls, number portability, telepowering, special number, connection through other networks)
VoIN Peer-to-Peer	NO	**NO**
VoIN IN&OUT	YES	YES (no emergency calls, number portability, telepowering, special number)
ToIP	YES	YS (no telepowering)

NGN Interconnection

- NGN interconenction is standardized by ETSI/TISPAN
- In ETSI standards, interconnection can be performed in two ways::
  - Service-oriented Interconnection (SoIx);
  - Connectivity-oriented Interconnection (CoIx).
- SoIx operates at the service layer, CoIx operates at the transport layer

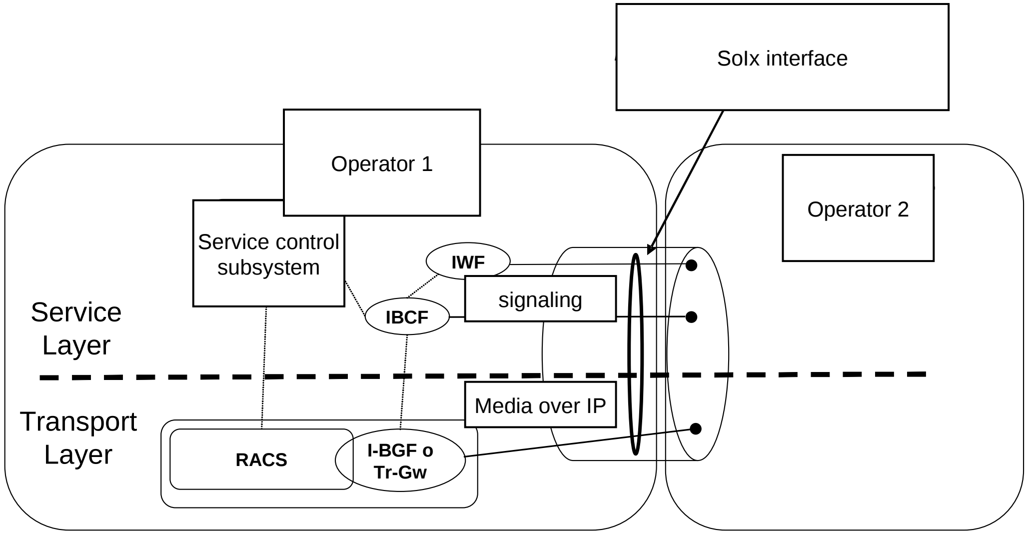


NGN Interconnection

- SoIx interconnection is the physical and logical interconnection between two administrative domains of different NGN operators. It allows operators to offers a complete service with QoS requirement in end-to-end fashion
- The CoIx interconnection operates basically at the IP layer, without considering service-level QoS requirements
- The CoIx interconnection may operate guaranteeing IP-layer service requirements

NGN Interconnection

- With reference to the interconnection for the delivery of ToIP services, the InterWorking Function (IWF) enables interworking of different signaling protocolssuch as SIP (Session Initiation Protocol) and ISUP (Integrated Services User Part)
- The Border Gateway Function (BGF) separates the two interconnected administrative domains, supporting security, QoS, call tracing, traffic logs
- The RACS (Resource and Admission Control Subsystem) function controls the usage of resources at the IP layer and it is responsible of QoS at the IP layer



Legenda
IBCF: Interconnection Border Control Function
I-BGF: Interconnection Border Gateway Function
Tr-GW: Transition GateWay in 3GPP
IWF: Interworking Function
RACS: Resource and Admission Control Subsystem

NGN Interconnection

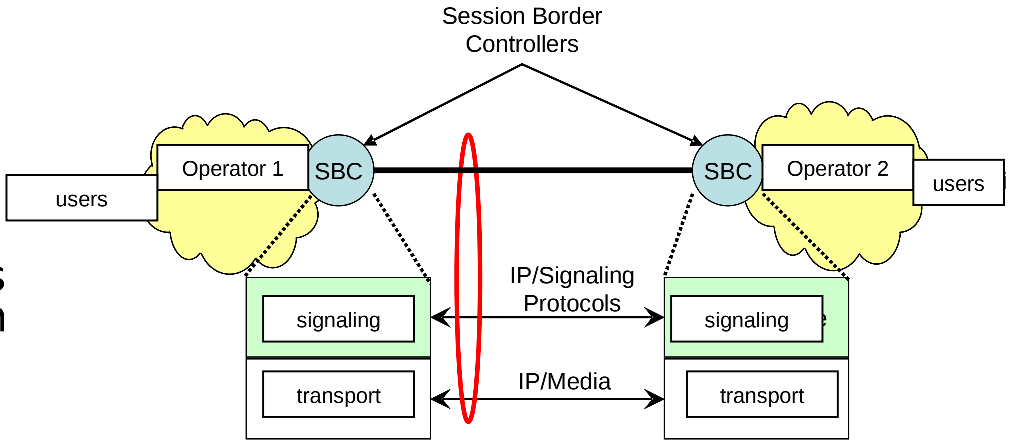
- The CoIx interconnection provides the connectivity allowing operators to let their customers reach external networks
- A typical example of CoIx interconnection is the peering IP service

NGN Interconnection

Interconnection service	Definizione	Esempi di applicazione	
		SoIx	CoIx
termination	Service requests are originated at the OLO/SP side and terminated onto customers of the interconnected operator	Telephone termination e videotelephony Termination of video-streaming session based Messaging (SMS, MMS...)	N.A.
collection	Service requestsof customers are forwarded to OLO/SP	Communication to non-geographic numbers Carrier Selection & Carrier Pre-selection Internet Dial-up	N.A.
transit	Service requests of OLO/SP transit through the interconnected operator	Transit of telephony & video telephony	N.A.
IP transit	IP traffic of OLO/SP transit through the interconnected operator	N.A.	IP transit to Peering domains
access	IP traffic of customers are forwarded to OLO/SP	N.A.	Bitstream Leased Lines (Terminating)
IP transport	IP traffic of OLO/SP between two remote points transits through the interconnected operator	N.A.	Leased Lines (trunk) VPN interconnection

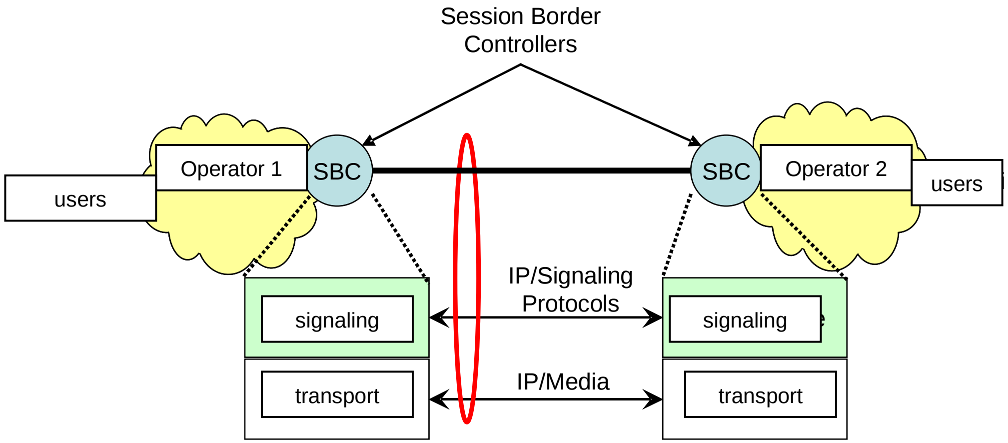
Solx requirements

- Signaling requirements:
  - Interoperability of signaling and service identification
- Requirements on codecs:
  - A set of codecs must be supported: at least G.711 bur are recommended Adaptive Multirate (AMR), G.729A and Enhanced Variable Rate Code (EVCR)
- Authomatic selection of codecs must be supported, the system must be able to scale down to the lowest quality codec involved in the session. Audio transcoding must be supported.
- Video codecs: at least H.263 and H.264



SoIx requirements

- routing:
  - Service-based routing must be supported
- security
  - Lawful interception, authorization, authentication, access control, data integrity, privacy
- Billing and accounting
  - Logs, traffic reports, billing generation, Charging Data Record (CDR).
- QoS & SLA
  - Resource reservation for QoS-aware sessions
- Connection Admission Control



Multimedia streaming

Media streaming

- Media streaming is the distribution of audio and video contents, in general synchronized
- The basic feature of media streaming is that content is used while it is transferred through the network, instead of first downloding the content and then playing it on the user’s client
- This is the most important difference between media streaming and the classic file transfer service

Media streaming

- Voice/audio and video codecs play an important role in media streaming
- Some codecs provide just one transmission speed, while layered codecs and multiple-description codecs provide a set of multiple transmission speeds, usually corresponding to different quality of the media
- Layered codecs logically divide the media stream into a set of separate substreams, called layers
- The user must receive the basic layer to be able to reproduce the content, at the lower quality level
- Additional layers can provide better quality, if the user’s access bandwidth is enough to transport them
- In this way, layered codecs are flexible as it it possible to adapt media quality to user’s resources
- Disadvantage: usually a layered codec consumer more bandwidth than a non-layered codec, given an equal quality level

Media streaming: communication costs

- Media transport costs are a relevent issue for carrier-grade multuimedia streaming
- Cost targets vary: a reasonable figure is that the cost of the transport of an average film coded with MPEG2@5 Mbit/s should not exceed 1 USD
- This results into about 0.00023 USD/Mbyte for a content with average duration
- In order to reduce costs of transport, architectural solutions such as chaching can be adopted
- Also multicasting is an efficient solution

Media streaming

- A content distribution service adopting media streaming as a transport technology can be providen in two main fashions:
  - Pull: the user explicitly requires a specific content
  - Push: contents are provided, through a distribution channel, according to a schedule
- A second characterization of a content distribution service is concerned with delay:
  - Live events (for example a soccer match) require very small delays, on the order of 1 s
  - Films are much more tolerant, as delay is concerned

Playout buffer

- In the user’s client, a part of the content is accumulated in the playout buffer before starting the replay of the content
- The playout buffer compensate the varying speed of the network
- The larger the playout buffer, the smaller is the frequency of buffer underflows, that degrade media quality
- However, delay increases as the size of the playout buffer grows
- A tradeoff is to be seeked between quality (small number of buffer underflows) and delay
- The tradeoff varies depending on the type of content