IP Telephony Access

IP Telephony Access, Transport, and Gateway Fundamentals

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Contents

+ IP Telephony Access,Transport, and Gateway Fundamentals

Where does the telecommunications industry's adventure in the intelligent network (IN) and Internet protocol (IP) space begin? This paper will provide a high-level discussion of the major issues and challenges, which result from the convergence of these two technologies.

+ Voice over IP: Why

Why is Voice over IP (VoIP) such a hot topic? First, there are compelling economies in transmitting voice-over-packet networks due to the nature of packet-based, open systems network implementations and with Moore's law. Secondly, relative traffic volumes of voice and data are intersecting with data, and in the near future, will become the dominant form of traffic on telephony networks. And with data traffic increasing at an exponential rate while voice is growing at a linear rate of only about three or four percent a year, in a relatively short timeframe, voice traffic will be insignificant compared to the volume of data on the network. In addition, the standards that will allow interoperability between the public-switched telephone network (PSTN) and the IP packet-switched network environments are coming to fruition.

What are the compelling economies in transmitting voice-over-packet networks? This can best be understood by analyzing the performance and cost (in bits per second per dollar invested) of circuit-switching and packet-switching technologies. While both are growing, clearly the economies are favoring packet technology. Again, this is a result of open systems technologies and Moore's law. (See *Figure 1*).

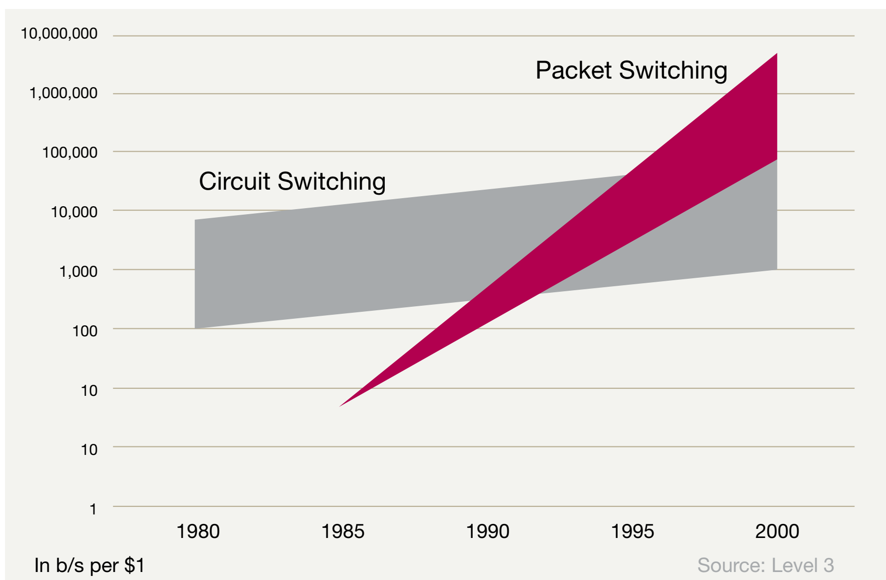

Figure 1:
PERFORMANCE/LOST CURVE

It is important to remember that the infrastructure and capital costs are only a very small fraction of overall costs to an enterprise introducing telephony capabilities. The capital itself is probably only about 20 percent of the cost, while the other 80 percent lies in the operational costs of the network. So *Figure 1* does not tell the whole story. But recently several carriers have claimed that IP-based packet technologies will also reduce their operations costs by 50 percent. If this is true, then clearly there are significant economic benefits of introducing a packet-based approach versus a circuit-based approach. By the way, this only applies to the Greenfield environment. The cost structure will be quite different in a legacy circuit-based networking environment.

The Yankee Group projected that in 2002 voice and data traffic volumes intersect. As stated earlier, data traffic is increasing at an exponential rate while voice is growing at a linear rate of only about three or four percent a year, and hence, voice traffic will be insignificant compared to the volume of data on the network. However, about 80 percent of the revenue come from voice services and only 20 percent from data services. This creates a very interesting situation where data creates the greatest volume of traffic, but voice creates far more revenue.
Over the next five years that situation should play out in some very interesting unpredictable ways (see *Figure 2*).

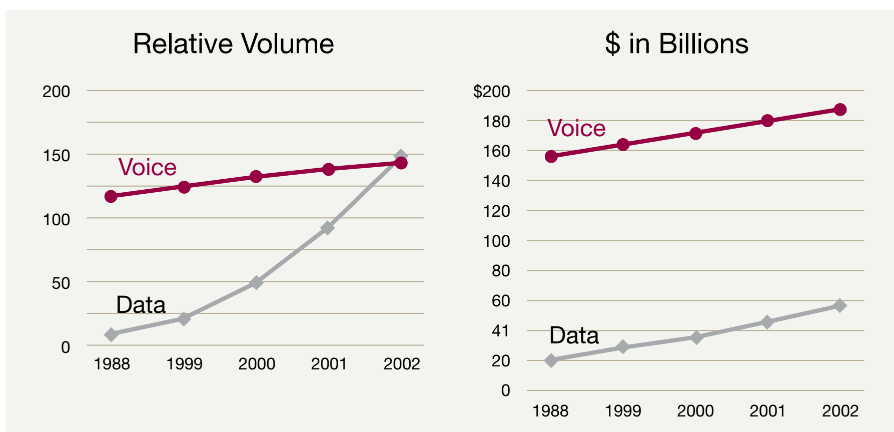

Figure 2:
TRAFFIC AND REVENUE GROWTH, 1998-2002

+ IP Telephony Networks

In an IP telephony network, it is critical that the IP networks inter-connect with the PSTN. To accomplish this, PSTN-IP architectures have been defined which allow for seamless interoperability. There are several different inter-connection schemas, one of which is shown in *Figure 3*. In this scheme, the class 5 SSP connects to the IP media gateway, which is essentially a router with PSTN trunk interface capabilities. The media gateway essentially converts the PSTN DS-0 format into IP-based packets. At this point, the call enters the IP telephony network and is routed as an IP packet until the call/packet encounters the terminating media gateway, where it is translated back to DS-0 format and the call is terminated.

As shown in *Figure 3*, the IP telephony environment consists of the Media Gateway (MG), which was discussed briefly above, the Media Gateway Controller (MGC), which provides call control logic; the Gatekeeper, which is not unlike a service control point (SCP); the Signaling Gateway, which has IP connections and interfaces from the MGC; and Signaling System 7 (SS7) interfaces to the SS7 network.

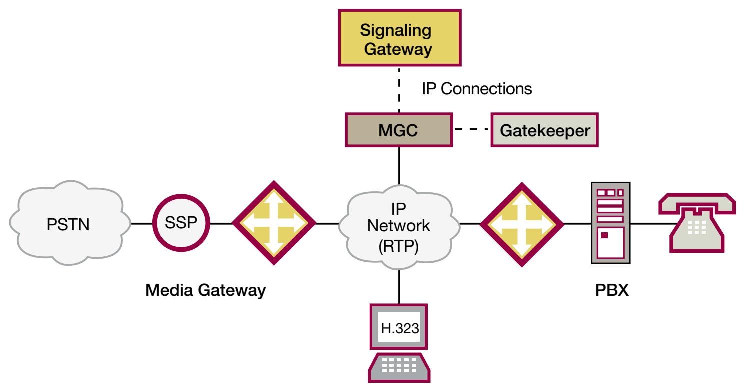

Figure 3:
IP TELEPHONY NETWORKS

IP telephony truly separates transport from the call control logic. This huge step in the evolution of the Intelligent Network (IN) that will lead to the realization of goals that were set out over 40 years ago in paper written by Irwin Dorros in 1955 in the Bell System Technical Journal which introduced the concept of SCPs into the PSTN. When Advanced IN (AIN) was introduced in 1984, the next step was taken as call control was disassociated from the transport layer. However, the switches/transport plane in the AIN architecture still had significant call control logic. In fact, hundreds of millions of lines of code were associated with the SSPs. It can be said that IP technology is the empowering agent for IN.

One element that is critical to the success of IP telephony is standards, not only standards for inter-connection to the PSTN, but standards for the inter-connection of the various IP-based components. The standardization of all these interfaces will lead to a new idea of what a carrier is. A carrier will no longer have to spend $5 million on an entire SSP. There will be carriers that buy just a media gateway, and then rely on other carriers for the call control logic in the media gateway controller (MGC). Other carriers might provide signaling gateway services to the MGC carriers. Standardization is creating huge opportunities and is changing the 100-year-old landscape of telephony.

The rest of this paper will focus on how this IP telephony network works. What are its basic elements? As shown in *Figure 3*, the IP telephony environment consists of the Media Gateway (MG), the Media Gateway Controller (MGC), the Gatekeeper, and the Signaling Gateway (SG). Fundamentally, what is really happening in IP telephony is that the central office is opening up, and the functionality of the SSP is being put into the servers mentioned above that have open systems standard IP-based interfaces between them. Simply put, if the media gateway, the MGC, the gatekeeper, and the signaling gateway were all combined in one box, the result would be called an SSP.

+ What Is a Media Gateway?

The media gateway is in the transport plane of the IP network architecture. It is network element that is inserted between the voice network (PSTN) and the IP network to perform the voice processing functions for translation between the voice world and the IP world. Beyond simple speech encoding and decoding, the media gateway must perform other telephony functions such as echo cancellation (usually performed by a toll network), playing announcements and call progress tones, call signaling, and logging and recording.

All these functions are normally found on a traditional Class 5 switch-line card. Basically the media gateway is terminating a DS-0 circuit, whether it’s embedded in a T-1 or OC 192 facility. It is taking the DS-0 channel and encoding it into IP packets. The first thing is does is encrypt and compress the DS-0 channel by taking the 64 kilobit per second channel down to 8 or 16 kilobits per second, depending upon the type of quality desired. The media gateway then packetizes the compressed information into an IP packet, places an IP address for the destination media gateway into the header, and sends it to the IP network.

So the media gateway is really the switching matrix of the central office (CO), but with no call control logic. The call control logic lies in the MGC and the gatekeeper.

The Media Gateway Controller

The MGC is the call control logic. It provides terminal requests and does call set-up and call control. It is important to note that since the IP carrier will have to inter-operate with the existing telephony infrastructure, it will have to provide the same level of service intelligence that is commonly associated with the current network. So, the IP carrier network will require access to SCP database services such as 8XX, Line Information Data Base (LIDB), Calling Name (CANM), and other Advanced IN services. As well, since the IP carrier will often be the n-1 provide, the MGC must be able to access Local Number Portability (LNP) database services. Hence, the MGC will need to trigger (or have a proxy trigger for it) at various points in the call to ask the distributed intelligence of the current and new networks how to route the call. The MGC will need the basic AIN call state machine with its core logic (or that of its proxy).

The Gatekeeper

The gatekeeper provides the services above and beyond basic call set-up and call control. These include the authentication, acknowledgement, and authorization (AAA) services. Interestingly, the IP world has the same type of mobility found in the wireless world: users in the IP world will plug their IP-based voice terminals into any IP outlet, and the carrier will have to authenticate and authorize that user as a valid IP telephony user. Messages will have to flow between the gatekeepers to recognize customers as authorized entities on the network and authorize the serving network to provide service to that customer. Roaming users in the wireless world must be authenticated and authorized as valid users, and there are a number of IS-41 messages that occur to accomplish that in the wireless world, which must be replicated in the IP space.

An issue of addressing arises between the PSTN space and the IP space. E.164 addresses are used for routing in the PSTN while the distributed switching IP network, on the other hand, uses IP addresses. So it is necessary to map E.164 addresses into IP addresses when transiting from an IP network to a PSTN network and vice-versa. Address mapping is another function of the gatekeeper.

+ IP-PSTN Call Set-Up Flow

Obviously an IP carrier will have to inter-connect to the existing telephony world, because without inter-connection, a carrier cannot even begin to offer service (other than in a closed private network environment). As a result, Integrated Services Digital Network (ISDN) User Part (ISUP) of SS7 is required for the MG to inter-connect with the SSP. The signaling gateway performs the SS7 over IP to SS7 protocol conversion (see *Figure 4*).

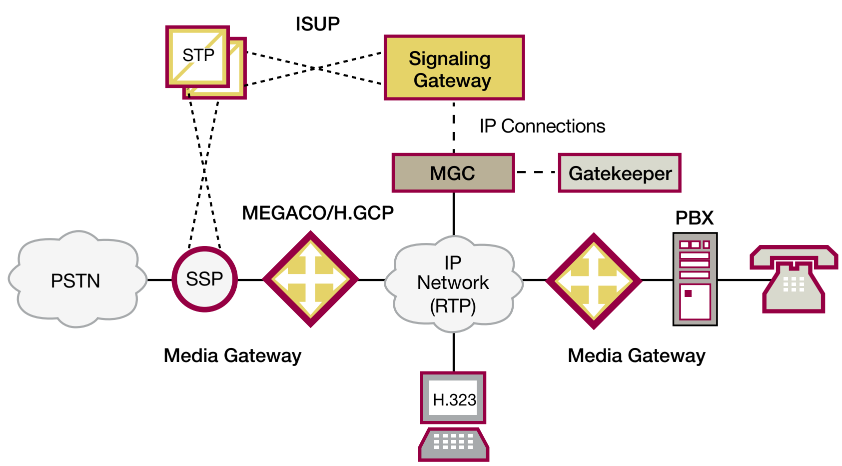

Figure 4:
IP TELEPHONY NETWORKS–THE NEED FOR ISUP

Consider a call that comes in from the PSTN, flows through the IP distributed network, and then terminates on the PBX. The PSTN will formulate an SS7 initial address message (IAM) that it sends to the SSP, that is, the terminating point or egress point to the IP network. The SSP will process the IAM and realize that the call is destined for the media gateway. The SSP will now need to establish a call to the media gateway and will send an IAM to the signaling transfer point (STP). The STP will read the destination address of the IAM and realize that it is destined for the Signaling Gateway. The signaling gateway interface is the regular SS7, ISUP over MTP-routed packet. The interface between the signaling gateway and the MGC is ISUP over IP. So the signaling gateway takes the SS7 IAM message, strips off the ISUP, encapsulates it in IP, and then sends the message over IP through the IP network to the MGC (see *Figure 5*).

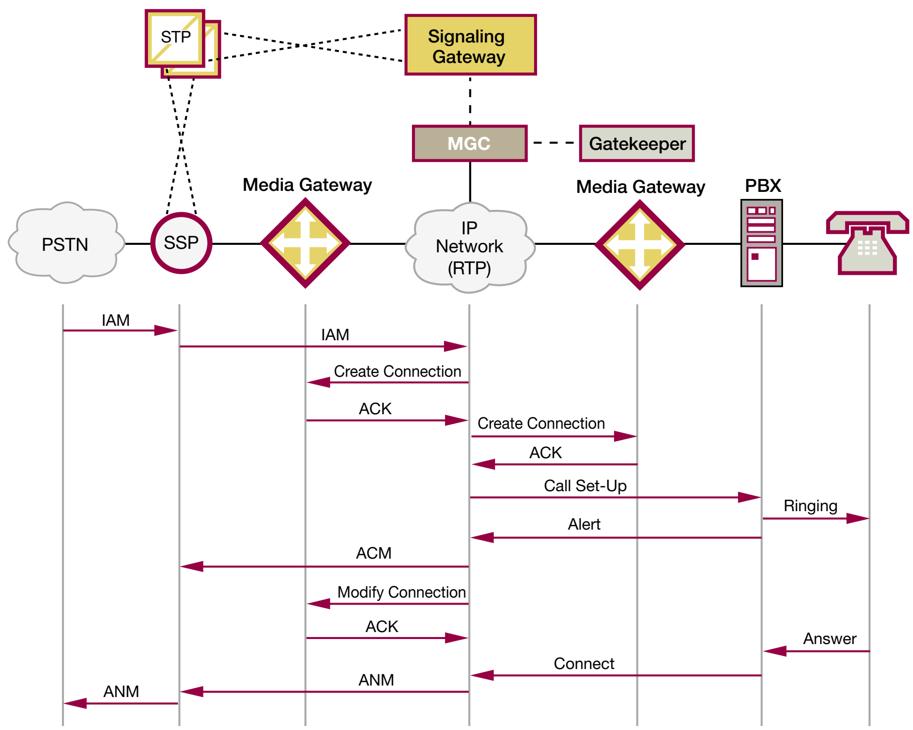

Figure 5:
IP-PSTN CALL SET-UP FLOW

Once the MGC has the ISUP message, it needs to communicate with the media gateway, much as a central office control module communicates with a line-trunk unit to control its connection during call set-up. So the MGC will send a “create connection” message to the media gateway, instructing the media gateway that there is a call request on a specific trunk group from the SSP (the circuit identification code or CIC). The media gateway will send an acknowledgement back to the MGC to say that everything is okay, and it is ready for the call.

In this example the MGC connects to two media gateways, the ingress and egress gateways, so it has visibility into the terminating media gateway through communications with the gatekeeper. The MGC will now use the functionality of the gatekeeper, informing the gatekeeper of the PSTN address of the terminating called party. The gatekeeper will return the IP address of the egress media gateway trunk, which connects to the PBX. The MGC controller now signals with the terminating media gateway and then sends a call set-up message to the PBX, including the called-party address information. The call is now complete.

The media gateways will also now have the address mapping information to put the appropriate IP terminating address on the compressed voice messages coming in off the SSP. The media gateway can then send that message on the IP network. The terminating media gateway will also have the ability to map the IP address into the E.164 terminating address as well as converting the IP voice packet into the DS-0 voice-bit stream.

Now that the connection has been created, a regular ISDN call set-up message flows to the PBX. The message rings the phone, alerts, and acknowledges, and then when the call is completed, an ISDN-connect message is sent to the IP network and an answer message is sent from the MGC to the signaling gateway, and again this is ISUP over IP. Then, the signaling gateway sends it to the STP and down to the SSP, where it is an ISUP over MTP message. The call is completed as normal.

+ IN and IP Telephony Networks

As mentioned before, not only will carriers be required to interoperate with the PSTN to perform call set-up, but they will also need to provide many of the advanced services available on the PSTN today. To compete on the local level, for instance, carriers will surely need caller ID with name (CNAM). CNAM has about 35 percent penetration in the United States. Customers will be very reluctant to disconnect their CNAM service to go with an IP carrier. As access to existing databases will still be required, the signaling gateway will not only have to handle the conversion of ISUP over MTP to ISUP over IP, but it will also have to handle transactional capabilities application part (TCAP) messages. Likewise, the MGC (or MGC proxy) will have to have a call model that will allow it to query out to the signaling gateway for TCAP-type services (see *Figure 6*).

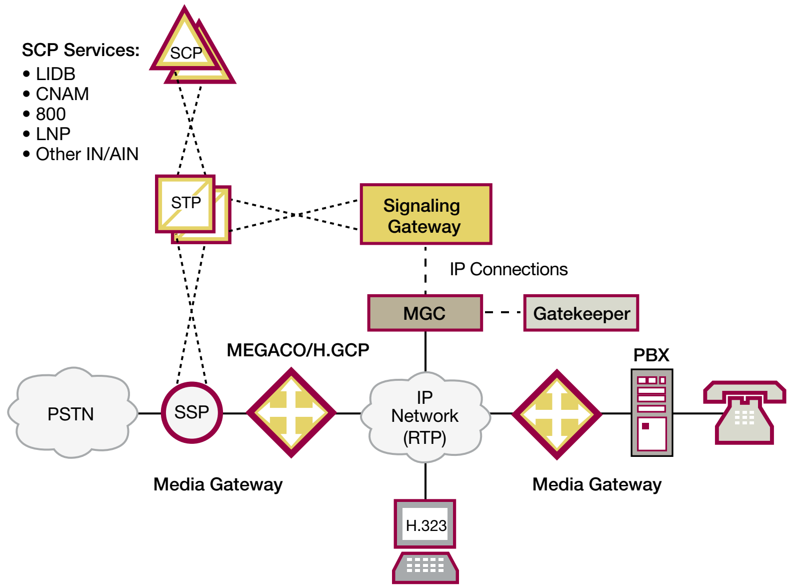

Figure 6:
IN AND IP TELEPHONY NETWORKS

+ Local Number Portability

A slightly different architecture diagram is required to understand local number portability (LNP) call-flows (see *Figure 7*).

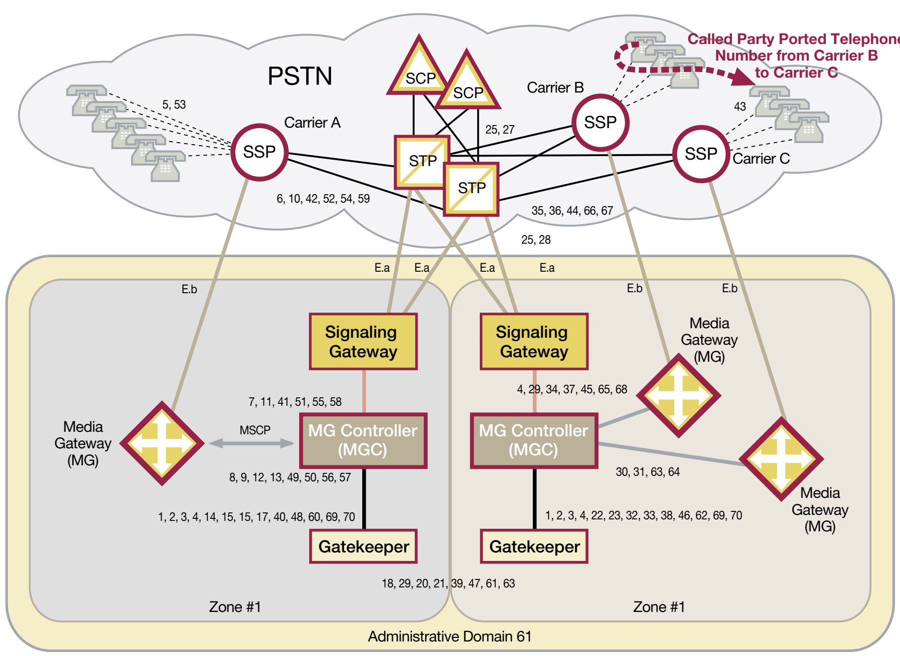

Figure 7:
LOCAL NUMBER PORTABILITY

The most important thing to notice in *Figure 7* is the numbering scheme, which makes it easier to walk through the following call-flow diagrams and note the signaling links and message flow. There are 40 or 50 steps in overall LNP call-flow, so it will not be discussed in detail, but in essence what happens is that the n-1 carrier is required to dip the LNP database. Hence, the MGC (or MGC proxy) will have an AIN call model that contains the NPA-NXXs that are ported, and when a number comes in that is in that NPA-NXX, it will have to trigger out to the LNP infrastructure to get the location routing number (LRN), and the call will then be routed successfully (see *Figure 8* and *Figure 9*).

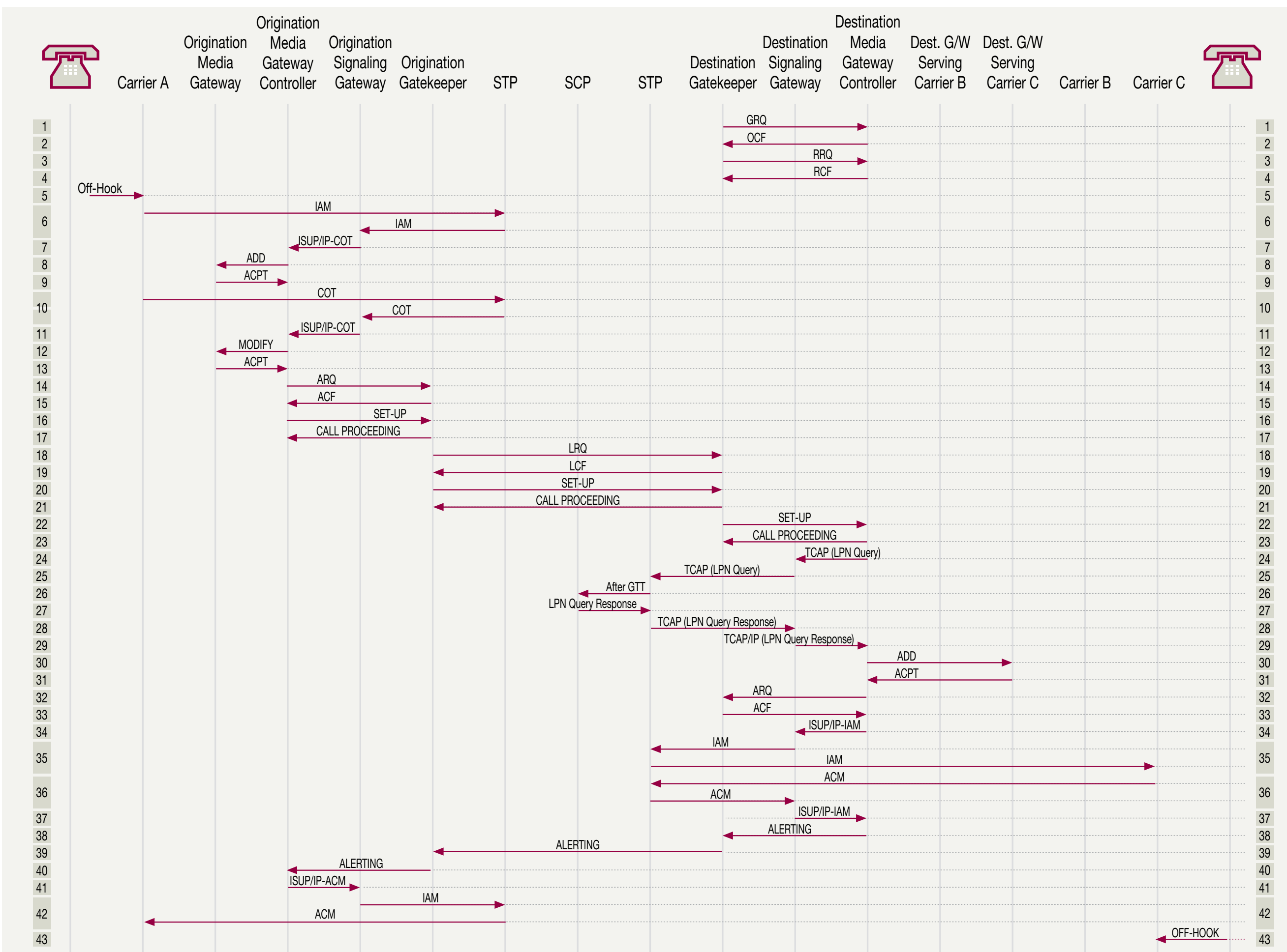

Figure 8 LPN CALL-FLOW

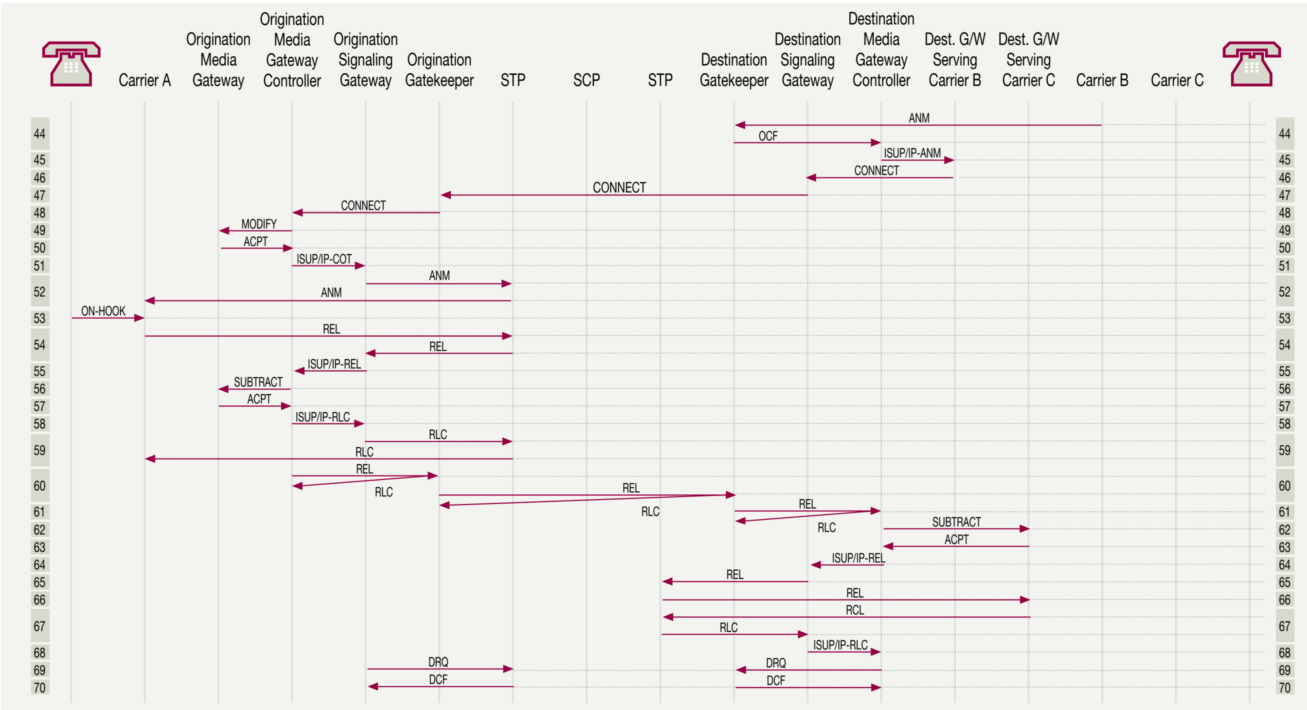

Figure 9 LPN CALL-FLOW CONTINUED

+ Interdomain Communication

In a highly interoperable environment where there are many IP carriers (and due to the cost-performance curve and the distributed nature of the IP telephony architecture there will be thousands of IP telephony networks deployed in the next five years), there will have to be a great deal of inter-carrier communication. An IP carrier, which has visibility into the address-mapping space for all of their terminations and originations, will not have visibility into the address-mapping schemes of other IP providers. So there will be a need for inter-carrier signaling and a need for neutral clearinghouse functionality between the carriers for billing information as well as address information. IP SCP services will include usage measurement and billing, authentication and fraud control, and inter-carrier, address-mapping number management.

+ Standardization

The architecture discussed above has been adopted by the ITU, the International Engineering Task Force (IETF), the IN Forum, and many standards bodies as the basic framework for how these IP-based entities (the media gateway, the MGC, the gatekeeper, the signaling gateway, the SS7 network, etc.) will fit together. Many complementary and competing standards have been defined. The number of standards suggests how many interfaces are required because, again, all these functions within the IP architecture have been distributed in an open systems environment. The underlying inter-connection element is an IP network to which everything now connects, which means that everything can talk to everything else. There are no proprietary connections as there are between a line trunk unit, the central controller administration module, and the trunk unit, where there is centralized control. Rather, this environment is based on open systems standards.

Many of the standards are evolving very quickly, so the following list is probably already outdated. Carriers are introducing protocols such as MGCP, SGCP, and MGCP/MEGACO. These are all call-control standards between the media gateway and the MGC. Then there is Q.2931, which is an asynchronous transfer mode (ATM) signaling protocol. There is also H.323, which has been a solid standard allowing IP terminals, multi-media terminals, voice, data, and video to communicate over an IP network to other H.323-compliant devices, whether terminals or gateways. Other standards include session initiation protocol (SIP) and SIP+, and ATM UNI 4.0.

+ Real-Time Transport Protocol

The real-time transport protocol (RTP) is the protocol used between media gateways. RTP carries data source and payload-type information and is itself carried inside of UDP. It handles packet sequencing, timing and synchronization, performance monitoring, and multi-media functions. RTP is really just the underlying transport plane, and the two media gateway controllers use it to talk to one another and successfully transport voice packets.

Part of the real-time protocol is the coding scheme. To send coded messages, both the originating and terminating gateways have to agree on a coding scheme. At the initiation of a session, information goes between the media gateways so that they can agree on a coding scheme. Surprisingly, in the G700 series coding schemes, as the bit rate goes up, the quality also goes up, and the complexity goes down (see *Figure 10*).

Algorithm	G.723.1	G.729	G.728	G.726	G.711
Rate (kbps)	5.3-6.3	8	16	32	64
Quality	Good	Good	Better	Better	Best
Complexity	Highest	High	Lower	Low	Lowest

Figure 10:
QUALITY OF CODING SCHEMES

+ IP Telephony Networks

The new IP networks that are evolving all inter-connect with the IP cloud, and importantly, that IP cloud is NOT the Internet. It will instead be an IP network that has guaranteed QoS capabilities to ensure appropriate timing and delay, so that communications will perform at the level of service expected by the customer (see *Figure 11*).

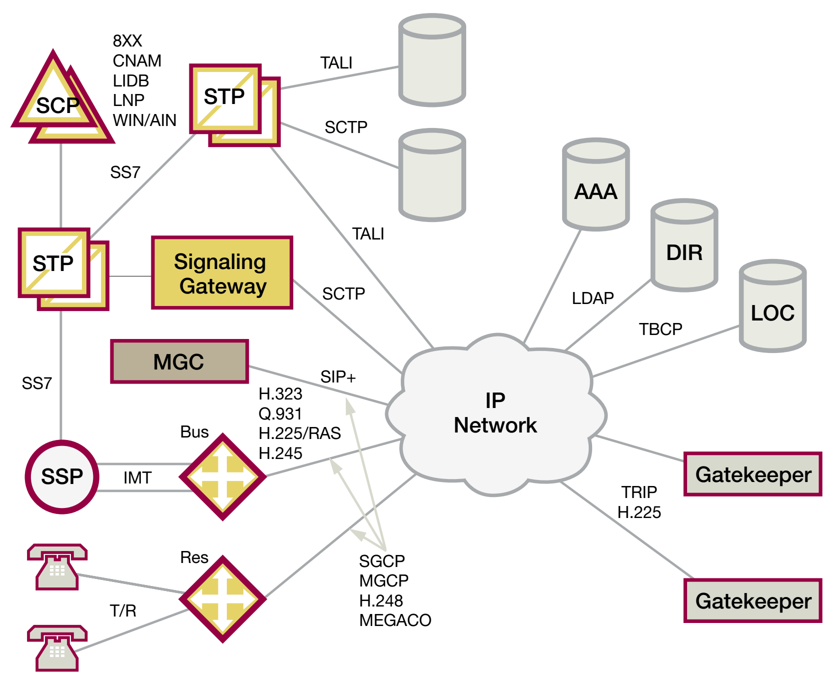

Figure 11:
TELEPHONY NETWORKS

All the various interfaces involved in IP telephony and PSTN interoperability can be seen on the left side of *Figure 11*. These include the traditional telephony network elements and interfaces including the SSP, SS7, Inter-Machine Trunks (IMTs), Tip-Ring (T/R), and database services. Also shown are the new IP telephony network elements and interfaces, including the signaling gateway interfacing between the SS7 network and the MGC via SCTP protocol, which is being defined in the IETF. SCTP is basically SS7, ISUP, and TCAP over IP with a middleware function to provide for MTP functions. Between the MGC and the media gateways, there is the entire H.323 family, Q.931, MGCP, Megaco, and the other protocols such as SIP. The media gateways communicate with one another via the reliable transfer protocol.

On the right side of *Figure 11* are depicted all the intelligence interfaces required to access databases, whether TALI or SCTP. Or, from the IP network, the interfaces are LDAP or TBCP, or TRIP and H.225 for the gatekeepers. All the documentation for these protocols is spelled out on the IETF and International Telecommunications Union (ITU) Web sites.

One of the things these architectures allows is a highly decentralized deployment, where there is a central call controller (MGC) and then media gateways highly distributed, all inter-connected via the IP network. The media gateways can be small devices that can terminate a few voice channels scaling to devices, which terminate thousands of voice channels. This makes it possible to start a telephone company in your basement. Of course, they can also scale to the point where the media gateway controller looks like a huge Class 5 or Class 4 SSP. So this architecture is highly scalable, highly decentralized, and highly distributed.

The highly distributed nature of IP telephony architectures introduces some important issues concerning SS7 signaling and routing and the impact on the North American numbering plan. This is because today a carrier deploys a centralized MGC and then deploys media gateways in every rate center. When they do that, each media gateway, under current numbering assignment rules, will need to be assigned an NPA-NXX block. Number pooling will solve many of the exhaustion problems. IP telephony is certainly a very different architecture that will have some interesting ramifications.

+ Internet Congestion Relief

What will be the initial application for this IP capability? While this paper has concentrated on IP voice, the first generation of MGC call control agents that are being deployed are being introduced to relieve Internet call congestion. Fifteen to 20 carriers have put in the architecture discussed above not for IP telephony but rather to relieve Internet congestion.

The Internet congestion problem comes about because the PSTN has been designed to handle phone calls with an average duration three minutes while the average Internet session lasts over 30 minutes. To support 10K Internet users during the Busy Hour (BH) requires 5K trunk groups (with 0.1 percent blocking). An equal number of voice calls requires 1K trunks (see *Figure 12* and *Figure 13*).

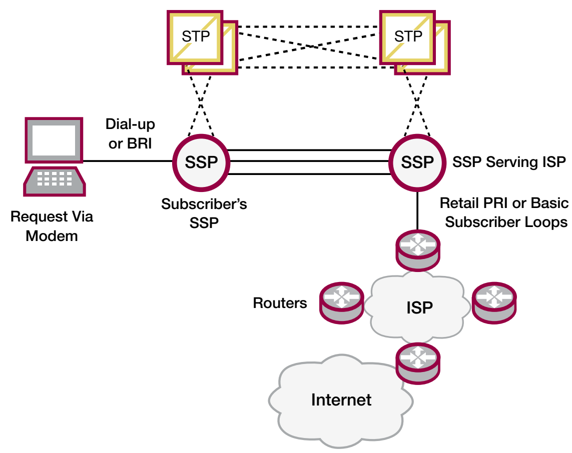

Figure 12

BEFORE INTERNET CALL DIVERSION ICD

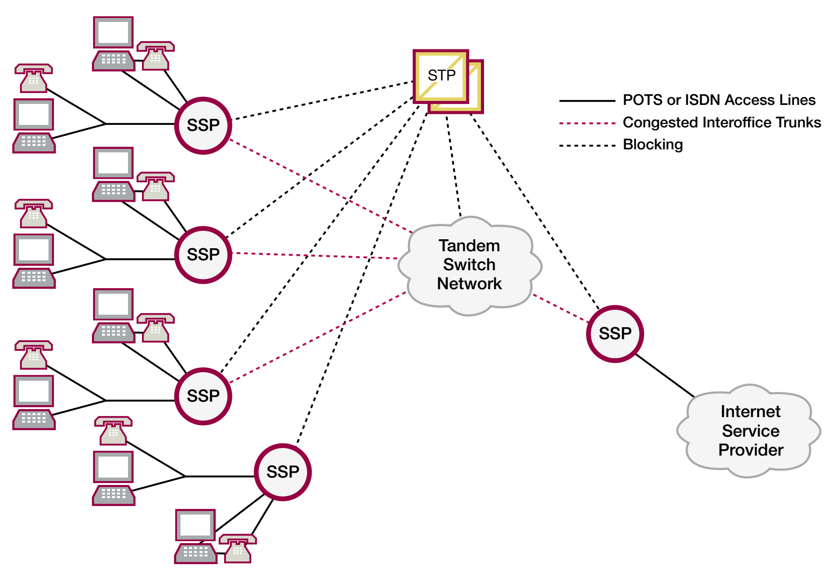

Figure 13

NETWORK CONGESTION CAUSED BY DIAL-UP INTERNET REQUESTS

Internet call diversion (ICD) alleviates these congestion problems (see *Figure 14* and *Figure 15*). Internet call diversion essentially allows direct trunk groups to connect the originating end-office (or Access Tandem) directly to the Internet, removing this traffic from the PSTN as soon as possible. This requires SS7 connectivity between the end-office and the Internet. The ICD device interfaces to the SS7 network and converts the SS7 ISUP messages to IP messages which are then sent to enhanced routers, and the call is completed. The ICD is basically the signaling gateway and MGC discussed above and the enhanced router is the media gateway.

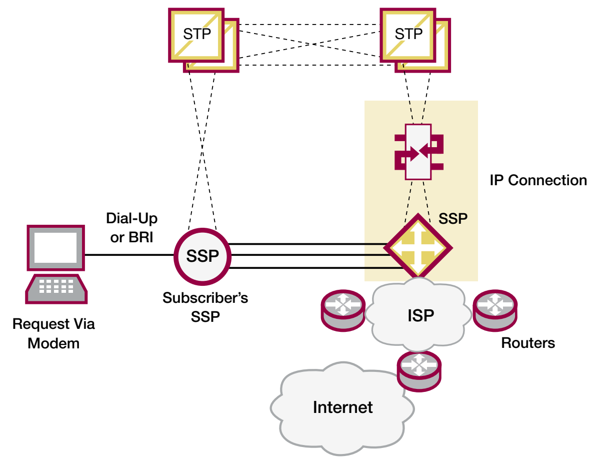

Figure 14
POST ICD DEPLOYMENT

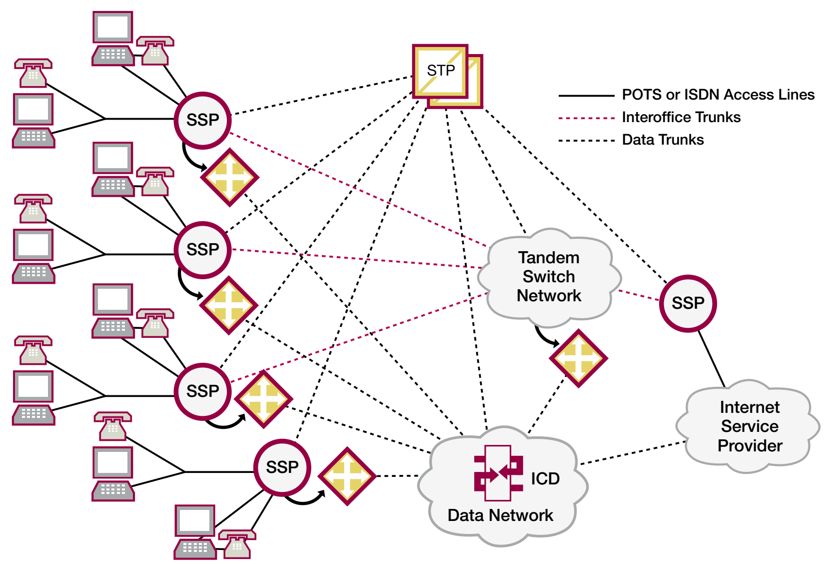

Figure 15
SS7 ELIMINATES NETWORK CONGESTION

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+ What Does This Really Mean

The most significant impact of IP telephony is the separation of call control from the network elements involved in call transport. The objective of the IN has been achieved: the transport and call control service application layers have been unbundled. It has taken a long time, but the industry is finally realizing the objectives set forth with the original SCP and IN/AIN concepts that have been unmet for many years.

This begs the final question: Where does the intelligence associated with communications go? Does it go to the edge (resulting in a dumb network) or to the core (resulting in a smart network)? The answer: It goes everywhere!