              Toward the PSTN/Internet Inter-Networking
                      --Pre-PINT Implementations

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Abstract

   This document contains the information relevant to the development of
   the inter-networking interfaces underway in the Public Switched
   Telephone Network (PSTN)/Internet Inter-Networking (PINT) Working
   Group. It addresses technologies, architectures, and several (but by
   no means all) existing pre-PINT implementations of the arrangements
   through which Internet applications can request and enrich PSTN
   telecommunications services. The common denominator of the enriched
   services (a.k.a. PINT services) is that they combine the Internet and
   PSTN services in such a way that the Internet is used for non-voice
   interactions, while the voice (and fax) are carried entirely over the
   PSTN. One key observation is that the pre-PINT implementations, being
   developed independently, do not inter-operate. It is a task of the
   PINT Working Group to define the inter-networking interfaces that

   will support inter-operation of the future implementations of PINT
   services.

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1. Introduction

   This document contains the information relevant to the development of
   the inter-networking interfaces underway in the Public Switched
   Telephone Network (PSTN)/Internet Inter-Networking (PINT) Working
   Group. It addresses technologies, architectures, and several (but by
   no means all) existing pre-PINT implementations of the arrangements
   through which Internet applications can request and enrich PSTN
   telecommunications services. The common denominator of the enriched
   services (a.k.a. PINT services) is that they combine the Internet and
   PSTN services in such a way that the Internet is used for non-voice
   interactions, while the voice (and fax) are carried entirely over the
   PSTN.

   The organization of the document is as follows. First, the basic
   terminology and a short "intuitive" description of the PINT services
   are provided. The rest of the information deals, in one way or the
   other, with the pre-PINT support of these services where they are
   used as a benchmark. Thus, an architectural overview common to all
   present solutions is presented. The flow of the document then
   divides into two streams: one is dedicated to the Intelligent Network
   (IN)-based solutions; the other explores alternative means (i.e.,
   CallBroker and Computer-Telephony Integration (CTI) approach). At
   this point, the emerging standards are explored, in particular, the
   Session Initiation Protocol (SIP), which promises an elegant solution
   to the PINT problem. Each of the above developments is addressed in a
   respective section. The final sections of the document contain the
   overall security considerations, conclusion, acknowledgments,
   appendix, and a set of references. The security section summarizes
   the PINT security requirements derived from the pre-PINT experiences
   and the appendix presents a tutorial on the PSTN, IN, and Call Center
   functions.

2. Terminology

   This document uses the following terminology:

   Authentication -- verification of the identity of a party.

   Authorization -- determination of whether or not a party has the
   right to perform certain activities.

   PINT Gateway -- the PSTN node that interacts with the Internet.

   User or Customer -- the person who asks for a service request to be
   issued. In the context of PINT Services, this person will use an
   Internet host to make his or her request. The term "user" is also
   used to describe a host originating the PINT service request on
   behalf of this person.

3. PINT Services

   This document addresses four services initially identified by the
   PINT Working Group and presently supported by pre-PINT
   implementations. These services are: click-to-dial-back, click-to-
   fax, click-to-fax-back and voice-access-to-content.

   Note that the word "click" should not be taken literally. It is
   rather used to point out that initiation of the related services
   takes place on the Internet, where point and click are the most
   prevalent user actions. In other words, a service request could
   originate from any type of IP-based platforms. There is no
   implication that these services must be implemented by a device
   within the PSTN or the Internet running a Web server.

   The common denominator of the PINT services is that they combine the
   Internet and PSTN services in such a way that the Internet is used
   for non-voice interactions, while the voice (and fax) are carried
   entirely over the PSTN. (An example of such a service is combination
   of a Web-based Yellow Pages service with the ability to initiate PSTN
   calls between customers and suppliers in a manner described in what
   follows.)

   Some of the benefits of using the PSTN are high quality of the voice,
   an ability to route the call to different locations depending on
   pre-set criteria (for example, time of the day, day of the week, and
   geographic location), outstanding security and reliability, and
   access to flexible, low cost, and secure billing and charging
   systems. The benefits of using the Internet are the uniform, well-
   defined, and widely-used interfaces available anywhere, anytime.

   Click-to-Dial-Back

   With this service, a user requests (through an IP host) that the PSTN
   call be established between another party and himself or herself. An
   important pre-requisite for using this service is that the user has
   simultaneous access to both the PSTN and Internet.

   One example of an application of this service is on-line shopping: a
   user browsing through an on-line catalogue, clicks a button thus
   inviting a call from a sales representative. Note that (as is the
   case with the all-PSTN Free-Phone, or "800", service) flexible

billing arrangements can be implemented here on behalf of the service
provider. In addition (and also similarly to the Free-Phone/800), the
PSTN could route the call depending on the time of day, day of week,
availability of agents in different locations, and so on.

Click-to-Fax

With this service, a user at an IP host requests that a fax be sent
to a particular fax number. In particular this service is especially
meaningful when the fax is to be sent to someone who has only a fax
machine (but no access to the Internet). Consider, as an example, a
service scenario in which a Web user makes a reservation for a hotel
room in Beijing from a travel service page containing hotel
information of major cities around the world. Suppose a specific
Beijing hotel chosen by the user does not have Internet connection
but has a fax machine. The user fills out the hotel reservation form
and then clicks a button sending out the form to the travel service
provider, which in turn generates a fax request and sends it together
with the hotel reservation form to the PSTN. Upon receiving the
request and the associated data, the PSTN translates the data into
the proper facsimile format and delivers it to the Beijing hotel as
specified in the fax request.

Click-to-Fax-Back

With this service, a user at an IP host can request that a fax be
sent to him or her. (Consider the user of the previous example, who
now requests the confirmation from the Beijing Hotel. Another useful
application of the service is when size of the information that a
user intends to get is so large that downloading it to the user's PC
over the Internet will require a long time and a lot of disk space.)

Voice-Access-to-Content

With this service, a user at an IP host requests that certain
information on the Internet be accessed (and delivered) in an audio
form over the PSTN, using the telephone as an informational
appliance. One application of this service is to provide Web access
to the blind. (This may require special resources--available in the
PSTN--to convert the Web data into speech.)

4. Architectural Overview

4.1 Public Switched Telephone Network

From an application perspective, Internet nodes are interconnected
directly, as shown in Figure 1. When two machines are to communicate,
they will have the address of the destination end system, and will

send network level datagrams, assuming that the underlying
infrastructure will deliver them as required.

```
                                      _____
                    __          _____/      \_____
                   [__]        /                  \
                   [----]-.-.-.-.    Internet        .-.
                              \_____          _____/  |
                           __  \__./      __     .
                          [__]   /       [__]    |
                          [----]-.        [----]-.
```
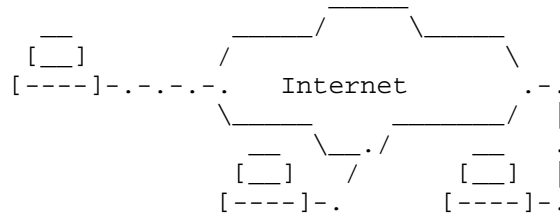
                   Key: .-.-. Internet Access Link

                              Figure 1

Where all nodes are on the same (broadcast) network, there is no need
for intervening routers; they can send and deliver packets to one
another directly. The Internet nodes are responsible for their own
communications requests, and act as peers in the communication
sessions that result.

This contrasts with the situation in the PSTN. There, the end systems
are configured as shown in Figure 2. The end systems tend to be
specific to a particular type of traffic, so that, for example, the
majority of terminals are dedicated to carrying speech traffic
(telephones) or to carrying facsimile data (fax machines). The
terminals all connect to Central Offices (COs) via access lines, and
these COs are interconnected into a network.
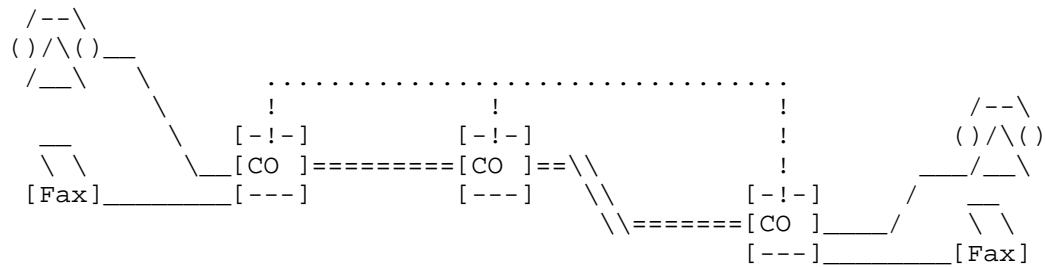
```
   /--\
  ()/\()__
  /__\   \      .................................
          \         !              !              !          /--\
    __      \     [-!-]          [-!-]            !          ()/\()
    \ \      \__[CO ]========[CO ]==\\           !        ___/__\
  [Fax]_____[---]          [---]   \\     [-!-]      /    __
                                        \\======[CO ]____/     \ \
                                        [---]_____[Fax]
```

   Key: ___   Access Lines
         ===   Trunk Links (inter-CO user data links)
         ...   Inter-CO signaling network links
         CO   Central Office (Telephone Exchange)

                              Figure 2

Communications between the terminals are all "circuit switched", so a dedicated synchronous data path (or circuit) needs to be placed between the end terminals for carrying all communications. Arranging for such a circuit to be made or removed (cleared) is the responsibility of the Central Offices in the network. A user makes a request via his or her terminal, and this request is passed on to the "local" Central Office. The relationship between the terminals and the local Central Offices to which they are connected is strictly Client/Server.

The COs are interconnected using two different types of connections. One of these is called a trunk connection (shown as a double line in the above figure) and is used to carry the data traffic generated by the terminals. The other connection acts as part of a separate network (and is shown as a dotted line in the above figure). This is the signaling network, and is used by the Central Offices to request a connection to be made between themselves and the destination of the required circuit. This will be carried across the trunk link to the "next" Central Office in the path. The path, once in place through the PSTN, always takes the same route. This contrasts with the Internet, where the underlying datagram nature of the infrastructure means that data packets are carried over different routes, depending on the combined traffic flows through the network at the time.

The call set up process can be viewed as having two parts: one in which a request for connection is made, and the other in which the circuit is made across the PSTN and call data flows between the communicating parties. This is shown in the next pair of figures (3a and 3b).

```
                        /--\
                       ()  ()
                        --____
                       /++\    \
                      /----\    \
                        A    \    [-!-]
                              \->[CO ]
                                 [---]
                    Time = 13:55
```

                            Figure 3a

    Key: ___   Access Lines
         ===   Trunk Links (inter-CO user data links)
         ...   Inter-CO signaling network links
         CO   Central Office (Telephone Exchange)

```
     /--\
    ()  ()
     --          ...............................
    /  \<---      ^                !                 !           /--\
   /----\   \     !                v                 !          ()  ()
     A'    \   [-!-]            [-!-]                !            --
            \__[CO ]========[CO ]==\\                v        ->-/  \
              [---]           [---]   \\         [-!-]       /  /----\
                                       \\======[CO ]____/        B'
    Time = 14:00                                [---]
```

                                Figure 3b

Figure 3 shows a particular kind of service that can be provided;
call booking. With this service, a request is sent for a connection
to be made between the A and B telephones at a specified time. The
telephone is then replaced (the request phase is terminated). At the
specified time, the CO will make a connection across the network in
the normal way, but will, first, ring the "local" or A' telephone to
inform the user that his or her call is now about to be made.

For more complex services, the requesting telephone is often
connected via its "local" CO to a Service Node (SN), where the user
can be played prompts and can specify the parameters of his or her
request in a more flexible manner. This is shown below, in Figures
4a and 4b. For more details of the operation of the Service Node (and
other Intelligent Network units), see the Appendix.

When the SN is involved in the request and in the call setup process,
it appears, to the CO, to be another PSTN terminal. As such, the
initial request is routed to the Service Node, which, as an end
system, then makes two independent calls "out" to A' and B'.

```
             /--\           [---]
            ()  ()          [SN ]
             --___          [|--]
            /++\  \           |
           /----\  \          |
                  \  \        |
            A      \   [|-!]
                    \->[CO ]
                       [---]
          Time = 13:55
```
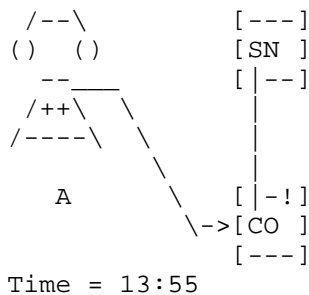
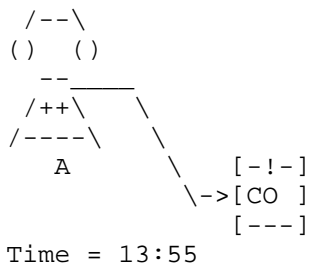   Figure 4a

```
   Key: ___     Access Lines
        ===     Trunk Links (inter-CO user data links)
        ...     Inter-CO signaling network links
        CO      Central Office (Telephone Exchange)
        SN      Service Node
```
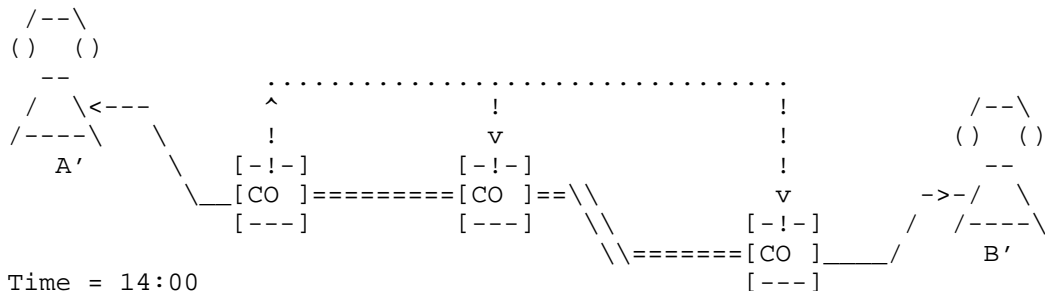
```
  /--\           [---]
 ()  ()          [SN ]
   --            [|--]                                         /--\
  /  \<--         |   ..............................         ()  ()
 /----\  \        |  ^               !               !         --
        \  \      | /               v               v        /  \
   A'    \   [|-!]          [-!-]           [-!-]    ->-/----\
           \--[CO ]         [CO ]           [CO ]    /
              [---]         [---]           [---]___/     B'
 Time = 14:00
```

Figure 4b

Note that in both cases as shown in Figures 3 and 4 a similar service
can be provided in which the B' telephone is replaced by an
Intelligent Peripheral (or an Special Resource Functional entity
within a Service Node), playing an announcement. This allows a "wake
up" call to be requested, with the Intelligent Peripheral or Service
Node Special Resource playing a suitable message to telephone A' at
the specified time. Again, for more details of the operation of the
Special Resources (and other Intelligent Network units), see the
Appendix.

4.2 Pre-PINT Systems

Although the pre-PINT systems reported here (i.e., those developed by
AT&T, Lucent, Siemens and Nortel) vary in the details of their
operation, they exhibit similarities in the architecture. This
section highlights the common features. Specific descriptions of
these systems will follow.

All of the systems can be seen as being quite similar to that shown
in the following diagram. In each case, the service is separated into
two parts; one for the request and another for execution of the
service. Figure 5 summarizes the process.

```
                           _____
            __       _____/      _____
           [__]     /                   \
          [-++-]-.-.>.-.    Internet        .-.-
                   _____          _____/    .
                       \___/               v
                                   [----]   .
                                   [PINT]-.-
                                   [----]
                                     %
                                     v
                                   [---]
                                   [SN ]
                                   [|--]


                              Figure 5a
```

```
     Key: CO     Central Office (Telephone Exchange)
          SN     Service Node
          PINT   PSTN/Internet Gateway
          .-.-.  Internet Access Link
          %%%    Gateway/Service Node Link
          ___    PSTN Access Lines
          ===    PSTN Trunk Links (inter-CO user data links)
          ...    Inter-CO signaling network links
```
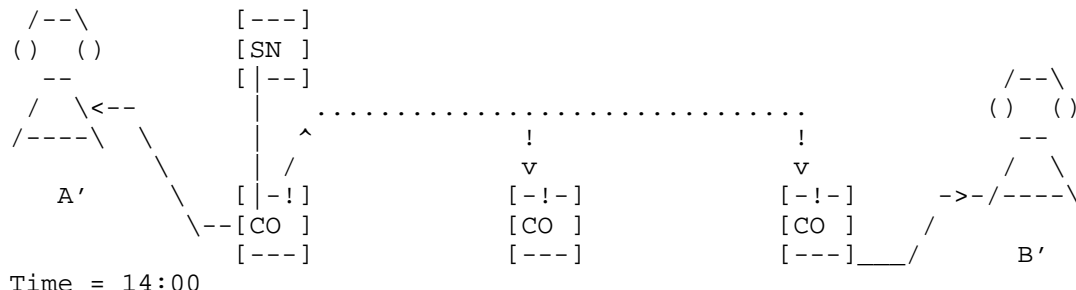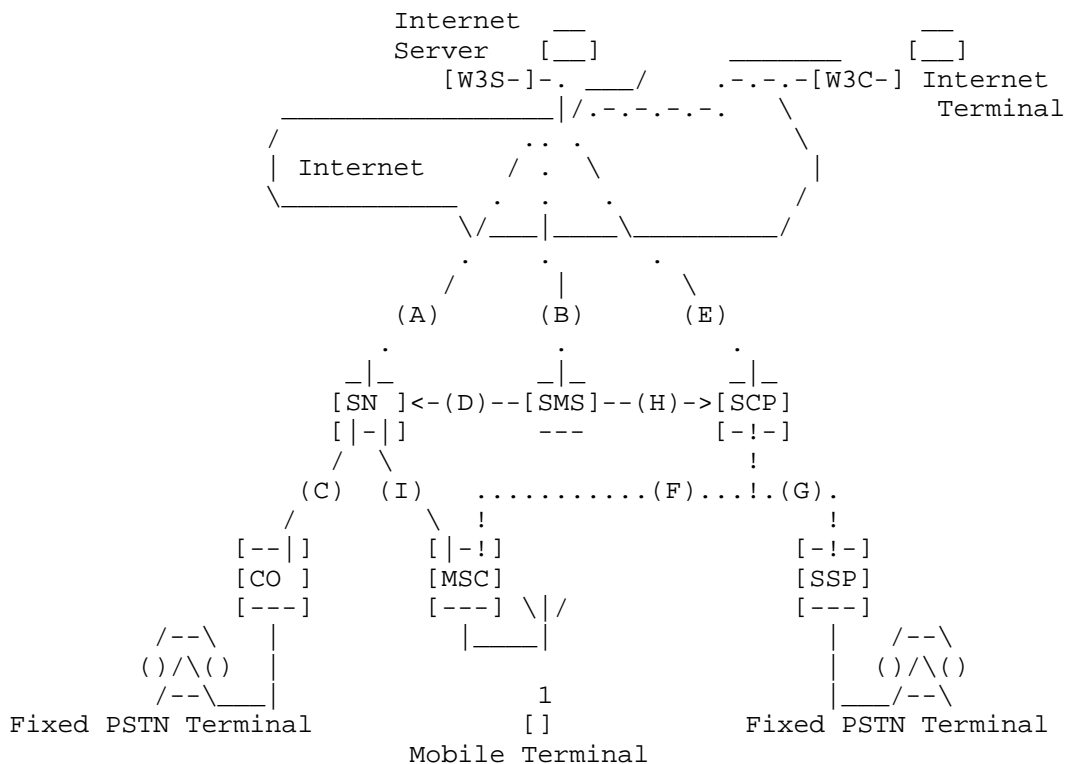
```
                           _____
            __       _____/      _____
           [__]     /                   \
         [----]-.-.-.-.    Internet        .-.-
                   _____          _____/    .
                       \___/                |
                                   [----]   .
                                   [PINT]-.-
                                   [-%--]
                                     %
                                     %
           /--\                    [-%-]
           ()  ()                  [SN ]
            --                     [|--]                      /--\
           /  \<--                   |   ....................  ()  ()
          /----\   \                 |   ^         !         !        --
                 \  |   /            v         v        / \
           A'     \  [|-!]         [-!-]       [-!-]   ->-/----\
                   \--[CO ]======[CO ]=====[CO ]   /
                     [---]       [---]       [---]__/     B'


                              Figure 5b
```

Comparing Figure 4a with Figure 5a, the differences lie in the way
that the information specifying the request is delivered to the
Service Node. In the PSTN/IN method shown in the earlier diagram, the
user connects to the SN from the telephone labeled A, with the
connection being routed via the CO. In the latter case, the request
is delivered from an Internet node, via the PINT gateway, and thence
to the Service Node over a "private" link. The effect is identical,
in that the request for service is specified (although the actual
parameters used to specify the service required may differ somewhat).

The figures depicting the respective service execution phases
(Figures 4b and 5b) show that the operation, from the IN/PSTN
perspective, is again identical. The Service Node appears to initiate
two independent calls "out" to telephones A' and B'.

The alternative systems developed by AT&T and by Nortel allow another
option to be used in which the PINT Gateway does not have to connect
to the PSTN via a Service Node (or other Intelligent Network
component), but can instead connect directly to Central Offices that
support the actions requested by the gateway. In these alternatives,
the commands are couched at a "lower level", specifying the call
states required for the intended service connection rather than the
service identifier and the addresses involved (leaving the
Intelligent Network components to coordinate the details of the
service call on the gateway's behalf). In this way the vocabulary of
the commands is closer to that used to control Central Offices. The
difference really lies in the language used for the services
specification, and all systems can use the overall architecture
depicted in Figure 5; the only question remains whether the
Intelligent Network components are actually needed in these other
approaches.

The following diagram (Figure 6) shows the interface architecture
involved in providing the kind of service mentioned above.

```
                       Internet  __                           __
                        Server   [__]        _____        [__]
                           [W3S-]-. ___/      .-.-.-[W3C-] Internet
                  _____|/.-.-.-.-.    \         Terminal
                 /                   .. .        \
                 | Internet      /  .  \           |
                 _____    .   .   .         /
                        \/___|_____/
                   .     .         .       .
                  /      |              \
                (A)      (B)            (E)
                  .         .          .
                _|_         _|_         _|_
              [SN ]<-(D)--[SMS]--(H)->[SCP]
              [|-|]         ---        [-!-]
              /   \                      !
            (C)   (I)   ...........(F)...!.(G).
            /        \   !                 !
         [--|]       [|-!]              [-!-]
         [CO ]       [MSC]              [SSP]
         [---]       [---] \|/          [---]
       /--\    |        |___|             |    /--\
      ()/\()   |                          |   ()/\()
      /--\___|             1              |___/--\
    Fixed PSTN Terminal         []       Fixed PSTN Terminal
                         Mobile Terminal
```

```
Key: W3S   HTTP (Web) Server
     W3C   HTTP (Web) Client/Browser
     CO    Central Office (Telephone Exchange)
     MSC   Mobile Switching Center (Mobile Network Telephone
           Exchange)
     SN    Service Node
     SSP   Service Switching Point
     SCP   Service Control Point
     SMS   Service Management System
     .-.-. Internet relationship
     ___   PSTN Access relationship
     ...   PSTN "core" signaling relationship
```

                       Figure 6

The interfaces are:

A   The interface over which Internet requests for service are
     delivered to the Service Node
B   The interface over which Service Management requests are sent
     from the Internet to the Service Management System
C   The interface over which the Service Node sends call control
     requests to a connected Central Office
D   The interface over which the Service Management System manages
     the Service Node
E   The interface over which Internet requests for service are
     delivered to the Service Control Point
F   The interface over which the Service Control Point sends service
     call control requests to the Mobile Switching Center
G   The interface over which the Service Control Point sends service
     control requests to the Service Switching Point
H   The interface over which the Service Management System manages
     the Service Control Point
I   The interface over which the Service Node sends service call
     control requests to the Mobile Switching Center

In practice, a number of the interfaces have very similar purposes to
one another. The means by which these purposes are achieved differ,
in that some of the interfaces (C and I) reflect access arrangements,
whilst others (F and G) imply a "core" signaling relationship.
However, it is possible to categorize them in terms of the "intent"
of messages sent across the interfaces.

For example, Interfaces A and E are similar; one of the main aims of
PINT work is to ensure that they are the same. Similarly, Interfaces
D and H imply similar actions and are likely to carry similar
messages. Interfaces C, F, G, and I are all used to request that a
call be initiated, albeit via access or core signaling relationships.

The interfaces can also be viewed in terms of the kind of components
that are involved and the bodies by which they are codified.
Interfaces A, B, and E are all going to be realized as Internet
Protocols. All of the others use existing protocols in the PSTN/IN.
Traditionally, these have been codified by different groups, and this
is likely to be the case in the PINT work.

The general arrangements for the different systems are shown below
(Figures 7, 8, 9, and 10). They differ in the details of their
configurations, but the main tasks they perform are very similar, and
so the overall operation is similar to the generic architecture shown
in Figures 5 and 6.

Key for following diagrams:

     Components:

W3C   World Wide Web Client
W3S   World Wide Web Server
WSA   Web Server "Back End Program" Interface (CGI or Servlet
        interface)
Srvlt   Servlet "back end" program/objects
FS   Finger Server
SCTPC   Simple Computer Telephony Protocol Client
SCTPS   Simple Computer Telephony Protocol Server
CBC   CallBroker Client
CBS   CallBroker Server
SSTPC   Service Support Transport Protocol Client
SSF   Service Switching Function
SCF   Service Control Function
SRF   Special Resource Function
CO   Central Office/ Public Telephone Exchange
SSP   Service Switching Point
SCP   Service Control Point
SR/I.IP Special Resource/ "Internet" Intelligent Peripheral
SMS   Service Management System
INAPAd Intelligent Network Application Part Adaptor
PktFlt Packet Filter (Firewall)
SNMPAg Simple Network Management Protocol Agent

     Protocols:

P0   HyperText Transfer Protocol
P1   HTTP Server <-> "Back End Program" internal protocol
P2   CallBroker Client <-> CallBroker Server protocol (AT&T system),
   or SCTP Client <-> Server protocol (Nortel system)
P3   PINT User Agent <-> PINT Gateway protocol
P4   Intra-Intelligent Network protocol (e.g., INAP)
P5   Proprietary (INAP-based) Gateway-> I.IP protocol
P6   Finger protocol
P7   Digital Subscriber Signaling 1 protocol
P8   Simple Network Management Protocol
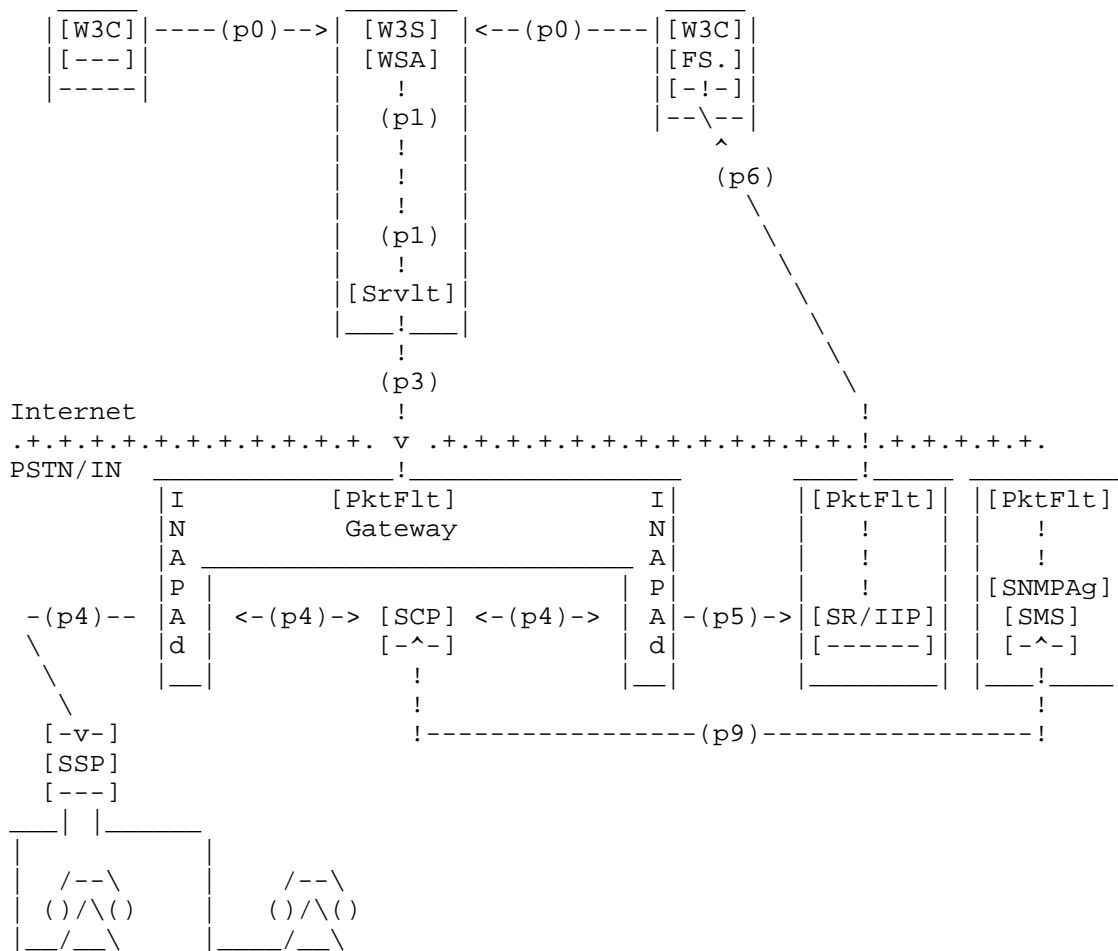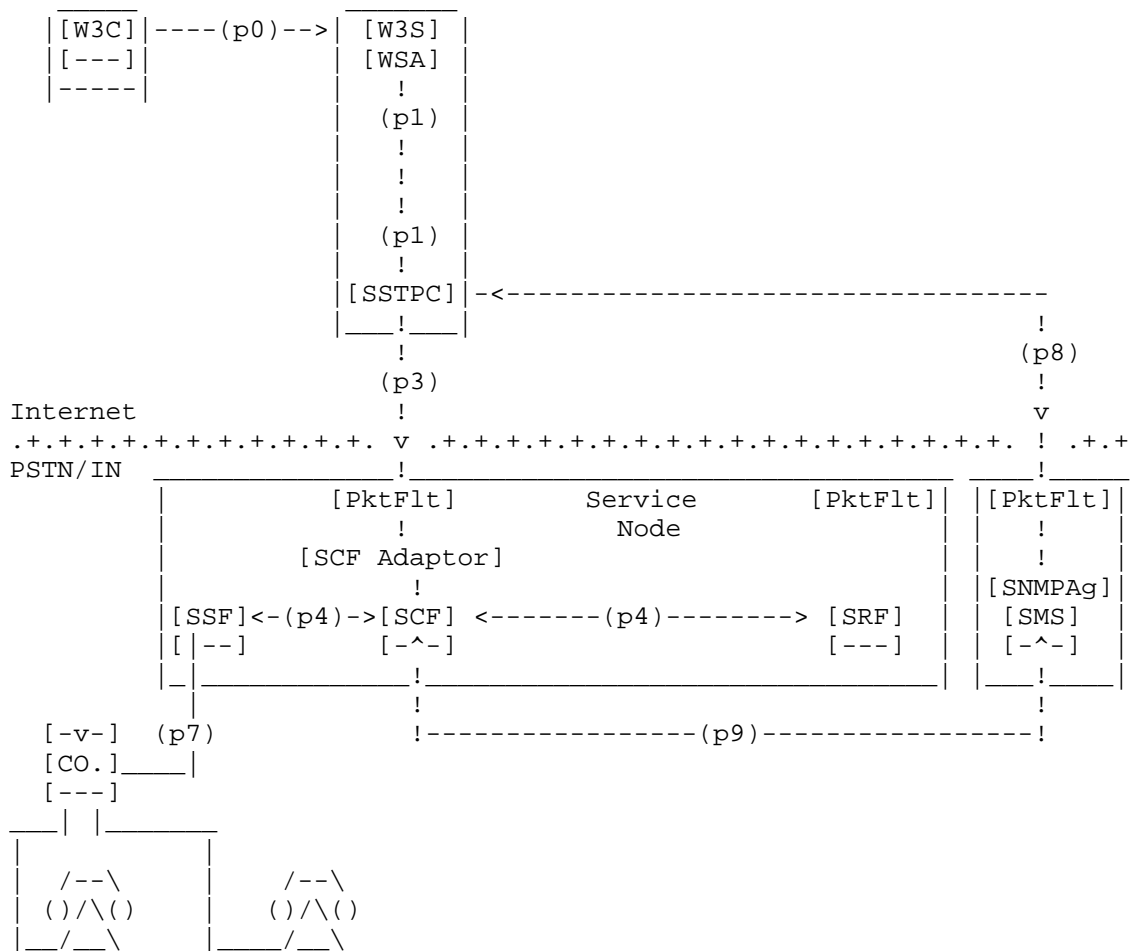P9   SMS <-> Service Control Point/Service Node protocol

```
    _____                 _____                 _____
   |[W3C]||----(p0)-->|  [W3S] |<--(p0)----|[W3C]|
   |[---]|            | [WSA]  |           |[FS.]|
   |-----|            |   !    |           |[-!-]|
                      |  (p1)  |           |--\--|
                      |   !    |              ^
                      |   !    |             (p6)
                      |   !    |               \
                      |  (p1)  |                \
                      |   !    |                 \
                      |[Srvlt]|                   \
                      |___!___|                    \
                          !                         \
                        (p3)                         \
   Internet                !                           !
   .+.+.+.+.+.+.+.+.+.+.+. v .+.+.+.+.+.+.+.+.+.+.+.+.!.+.+.+.+.
   PSTN/IN  _____!_____        ____!_____   _____
           |I          [PktFlt]          I|       |[PktFlt]|  |[PktFlt]|
           |N           Gateway          N|       |   !    |  |   !    |
           |A _____   A|      |   !    |  |   !    |
           |P |                         |  P|      |   !    |  |[SNMPAg]|
   -(p4)-- |A |  <-(p4)-> [SCP] <-(p4)-> |  A|-(p5)->|[SR/IIP]|  | [SMS]  |
      \    |d |          [-^-]          |  d|       |[------]|  | [-^-]  |
       \   |__|            !            |__|       |_____|  |___!____|
        \                  !                                        !
   [-v-]                   !----------------(p9)-----------------!
   [SSP]
   [---]
   ___| |_____
  |     |
  |  /--\      |    /--\
  | ()/\()     |   ()/\()
  |__/__\     |____/__\
```

Figure 7: The Siemens Web Call Center

```
           _____               _____
          |[W3C]|----(p0)-->|  [W3S] |
          |[---]|           |  [WSA] |
          |-----|           |   !    |
                            |  (p1)  |
                            |   !    |
                            |   !    |
                            |   !    |
                            |  (p1)  |
                            |   !    |
                            |[SSTPC]|-<--------------------------------
                            |___!___|                                 !
                                !                                   (p8)
                              (p3)                                   !
   Internet                    !                                     v
   .+.+.+.+.+.+.+.+.+.+.+. v .+.+.+.+.+.+.+.+.+.+.+.+.+.+.+.+.+. ! .+.+
   PSTN/IN  _____!_____  ___!____
           |        [PktFlt]       Service        [PktFlt]| |[PktFlt]|
           |                        Node                  | |   !    |
           |        [SCF Adaptor]                         | |   !    |
           |            !                                 | |[SNMPAg]|
           |[SSF]<-(p4)->[SCF] <-------(p4)--------> [SRF] | | [SMS]  |
           |[|--]       [-^-]                       [---]  | | [-^-]  |
           |_|_____!_____| |___!____|
            |            !                                      !
   [-v-]  (p7)          !----------------(p9)-----------------!
   [CO.]____|
   [---]
  ___| |_____
 |     |
 |  /--\     |    /--\
 | ()/\()    |   ()/\()
 |__/__\     |____/__\
```

Figure 8: The Lucent System

```
        _____                      _____
       |[W3C]||----(p0)-->|  [W3S]   |
       |[---]||                |  [WSA]   |
       |-----|                 |    !     |
                               |   (p1)   |
                               |    !     |
                               |[WS/CBS]  |
                               |[Adaptr]  |
                               |___!____|
                                    ^
                                  (p2)
        _____                 ___v____
       |[CBC]||                |  [CBS]   |
       |[---]||<---(p2)-->|  [---]   |-<----------------------------
       |-----|                 |___!____|                          !
                                    !                            (p8)
                                  (p3)                             !
  Internet                         !                              v
  .+.+.+.+.+.+.+.+.+.+.+. v .+.+.+.+.+.+.+.+.+.+.+.+.+.+.+.+.+. ! .+.+
  PSTN/IN _____!_____ ___!_____
         |        [PktFlt]      Service         [PktFlt]| |[PktFlt]|
         |            !          Node                   | |   !    |
         |      [SCF Adaptor]                           | |   !    |
         |            !                                 | |[SNMPAg]|
         |[SSF]<-(p4)->[SCF] <-------(p4)--------> [SRF] | | [SMS]  |
         |[|--]        [-^-]                     [---] | | [-^-]  |
         |_|_____!_____| |___!____|
          |            !                                      !
   [---]  (p7)         !----------------(p9)-----------------!
   [CO.]____|
   [---]
  ___| |_____
  |      |
  | /--\      |      /--\
  | ()/\()    |    ()/\()
  |__/__\     |____/__\
```

                    Figure 9: The AT&T System

```
  _____                   _____
 |[W3C]||----(p0)-->|  [W3S]   |
 |[---]||                 |  [WSA]   |
 |-----|                  |   !      |
                          |  (p1)    |
                          |   !      |
                          |[WS/   ]  |
                          |[ SCTPS]  |
                          |[Adaptr]  |
                          |___!____  |
                               ^
                             (p2)
  _____                  ___v___
 |[SCTPC]|                |[SCTPS]|
 |[-----]|  <-(p2)-->  |[-----]|-<--------------------------------
 |-------|                |___!___|                                !
                             !                                  (p8)
                           (p3)                                   !
Internet                    !                                     v
.+.+.+.+.+.+.+.+.+.+.+. v .+.+.+.+.+.+.+.+.+.+.+.+.+.+.+.+.+. ! .+.+.
PSTN/IN  _____!_____ ____!_____
        |          [PktFlt]      Service      [PktFlt]| |[PktFlt]|
        |             !            Node               | |   !    |
        |          [SCF Adaptor]                      | |   !    |
        |             !                               | |[SNMPAg]|
        |[SSF]<-(p4)->[SCF] <-------(p4)--------> [SRF] | | [SMS]  |
        |[|--]        [-^-]                      [---]  | | [-^-]  |
        |_|_____!_____| |__!____|
         |            !                                      !
    [---]  (p7)         !----------------(p9)-----------------!
    [CO.]____|
    [---]
   ___| |_____
  |         |
  |  /--\    |     /--\
  | ()/\()   |    ()/\()
  |__/__\    |____/__\
```
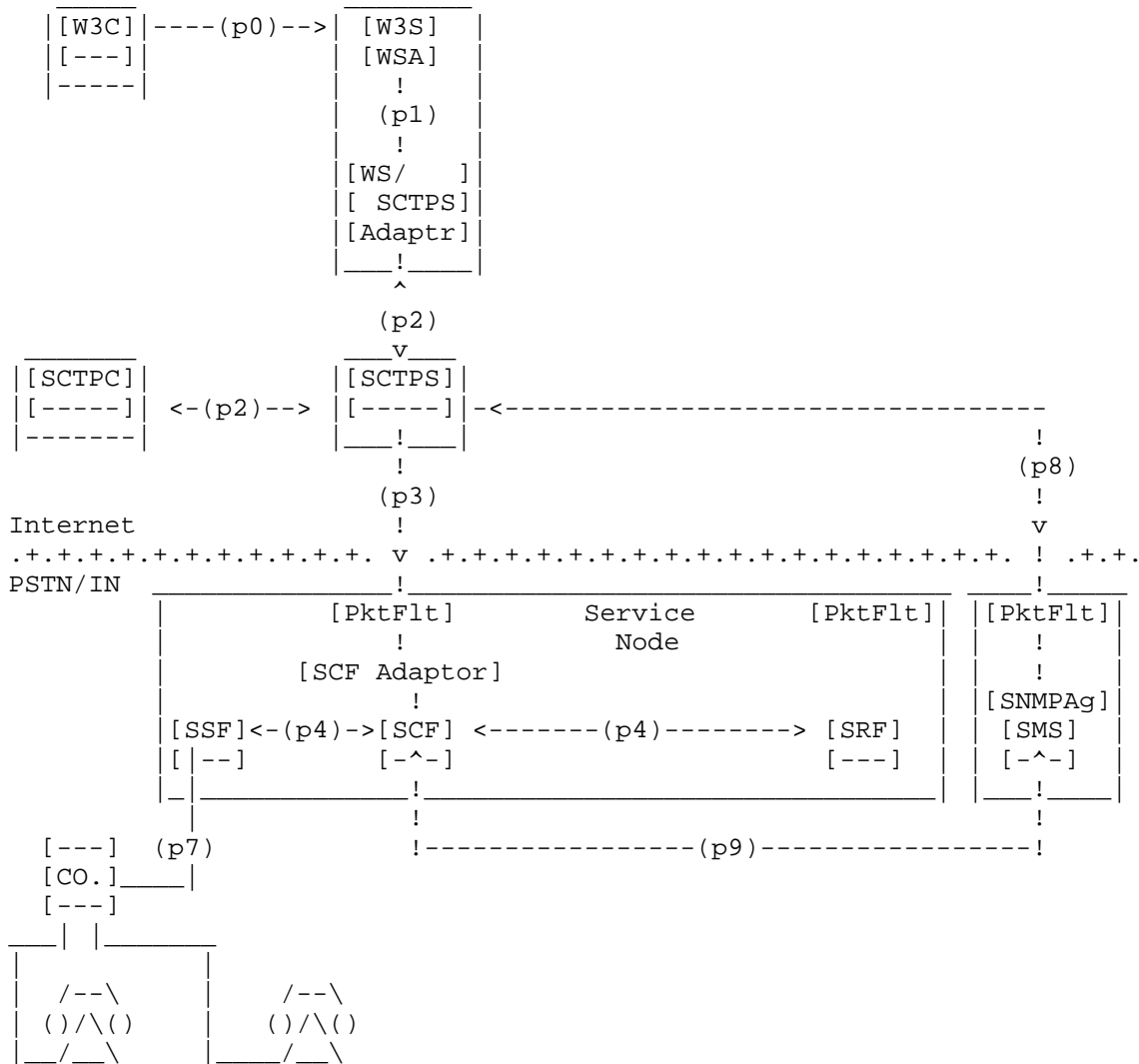
Figure 10: The Nortel System

   As these are independent systems developed by different groups, the
   names of the components, unsurprisingly, don't match. Some features
   are offered by one of the systems, while they aren't by others.
   However, there are a number of common features. All of the systems
   provide a Web-based interface (at least as an option), using "back
   end" programs to construct protocols to pass onwards to the
   Intelligent Network system.

Several Intelligent Network Functional Entities are combined into a
Service Node in the Lucent, AT&T , and Nortel systems, while in the
Siemens scheme they are separate units. However, this is not
particularly important for the provision of the services they offer.

The main difference lies in whether or not the SCF is "aware" of the
Internet interface and has been modified to be "complicit" in
supporting these Internet requests. The Siemens approach was to re-
use an existing SCP, providing a gateway function to translate as
needed. The Lucent system used a "lighter weight" SCF adapter to
terminate the Internet protocols, as the SCF was modified to support
the Internet interface directly.

The AT&T CallBroker and Nortel SCTP Servers introduce an intermediate
protocol (labeled p2) that allows an alternative to the Web based
interface supported by the others. This protocol matches the
"CallBroker Client API", or the "SCTP Client API". These options
provide for a bi-directional protocol, with indications sent from the
Call Broker or SCTP Server to the Client as needed. This is not
easily possible using an HTTP-based scheme (and in the Siemens case,
a dedicated Finger client/server pair was used to emulate such an
interface)

The protocol between the Internet server and the Intelligent Network
(labeled p3 in the above diagrams) differs in each of the systems.
One of the main aims of future work will be to develop a common
protocol that will support the services offered, so that the p3
interface will allow different implementations to inter-operate. In
the Lucent, Siemens, and Nortel systems, this was an "internal"
protocol, as it was carried between entities within the Service Node
or Gateway.

Other contrasts between the systems lie in the support for Internet
access to Service Management, and access to the Internet by Special
Resources. Internet Management access was most developed in the
Lucent system, in which a Simple Network Management Protocol (SNMP)
agent was provided to allow inter-operation with the SMS controlling
the Service Node. In the Siemens scheme, the SMS had no direct
Internet access; any management actions were carried out within the
normal PSTN management activities. As for Internet access to special
resources, this was only required by the Siemens system as part of
its support for Call Center agent notification. Equivalent
functionality would be provided in the AT&T and Nortel systems as
mentioned above, and this would in turn be associated with event
notifications being sent as part of their (p3) Internet/IN protocol.
These differences reflect the different emphases in the products as
they were developed; again, future work will have to ensure that
common protocols can be used to support the chosen services fully.

5. IN-Based Solutions

5.1 The Lucent System

   Figure 11 depicts the overall interconnection architecture of the
   Lucent prototype in support of the four PINT services. The IN-based
   architecture utilizes the Service Node and Service Management System
   in addition to the Web server, which enables Web-based access to the
   PINT services. This section summarizes the roles of these elements
   (complemented by a click-to-dial-back service scenario), outlines the
   interfaces of Web Server-Service Node and Web Server-Service
   Management System (i.e., the interfaces A & B), and addresses the
   common security concerns.

5.1.1 Roles of the Web Server, Service Node, and Service Management
      System

   Web Server

   The Web Server stores the profiles of content providers as well as
   pre-registered users. The content provider profile contains
   information such as content provider ID, telephone number, and fax
   number. In addition, the profile may also include service logic that
   specifies, for example, the telephone (or fax) number to be reached
   based on time of the day, day of the week, or geographical location
   of the user, and the conditions to accept the charge of the calls.

   Similar to the content provider profile, the pre-registered user
   profile contains information such as user name, password, telephone
   number, and fax number. The last two pieces of information can also
   be linked to time of the day and day of the week so the user can be
   reached at the appropriate telephone (or fax) number accordingly.

   Service Node

   Situated in the PSTN, the SN, like the SCP, performs the service
   control function [1, 2, 3]. It executes service logic and instructs
   switches on how to complete a call. The SN also performs certain
   switching functions (like bridging of calls) as well as a set of
   specialized functions (like playing announcements, voice recognition
   and text-to-speech conversion).

   Service Management System

   The SMS performs administration and management of service logic and
   customer-related data on the SN. It is responsible for the
   replication of content provider profiles and provision of these data
   on the SN. These functions are non-real time.

```
 Web Users
                                    _____
   O ------------------------- | Internet |------------------
                                ------------                |
                                                            |
                                                            |
   ----------------      -------------              ------------
  | Service Node |   D   | Service    |      B      |Web Server|
  |     (SN)     |-----------| Management |--------------|          |
  |              |       |System (SMS)|             |          |
  |              |    A   -------------              |          |
  |              |-------------------------------------|          |
   ----------------                                   ------------
      |          |
      |          |
      | I        | C
      |          |
   ----------- ---------
  |Mobile    | |Central|
  |Switching| |Office |
  | Center   | ---------
   -----------       |
      |          |
      |          |
      O          O

      Mobile      Wireline PSTN
      Users       Users
```
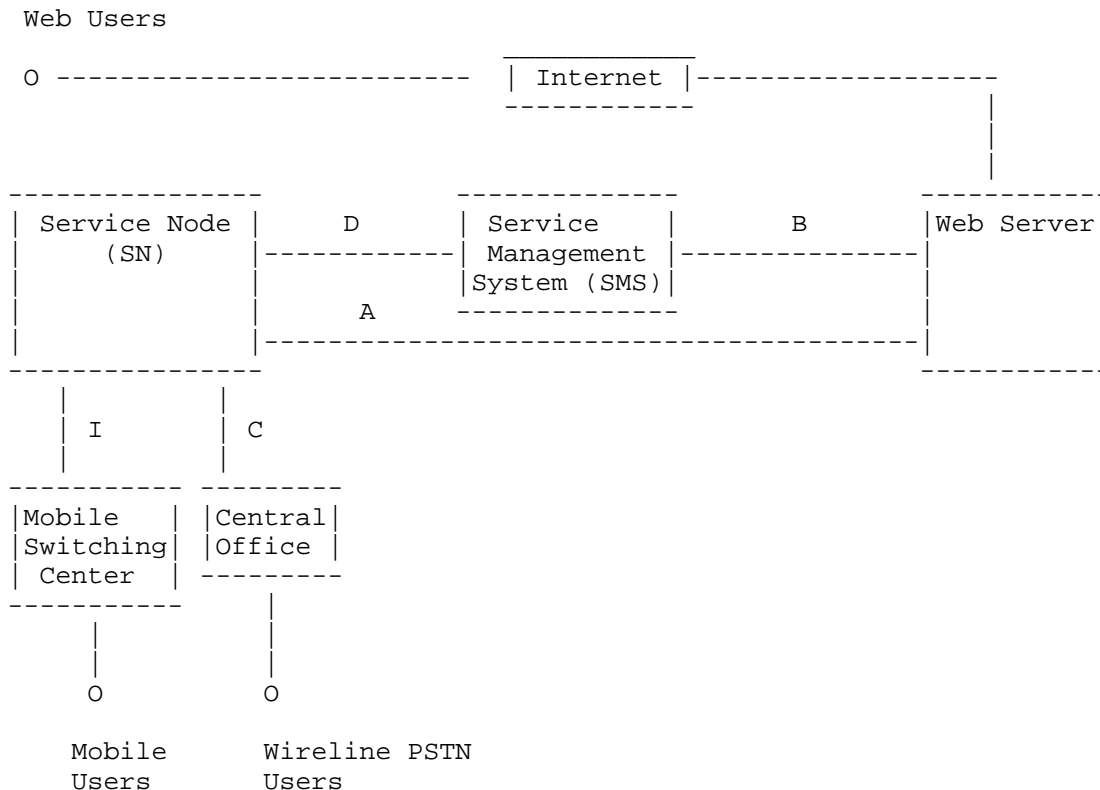
        Figure 11: Overall Interconnection Architecture of the Lucent System

5.1.2 A Click-to-Dial-Back Service Scenario

   A Web user, who has simultaneous access to the Web and telephone
   services (this can be achieved, for example, by having an ISDN
   connection), is browsing through a sales catalogue and deciding to
   speak to a sales representative.

   When the Web user clicks a button inviting a telephone call from the
   sales office, the Web Server sends a message to the SN over the A
   interface, thus crossing the Internet-to-PSTN boundary. By matching
   the information received from the Web Server with the content
   provider profile that had been previously loaded and activated by the
   SMS over the D interface, the SN recognizes the signal.

   At this point, the SN calls the Web user. The user answers the call,
   hears an announcement, e.g., "Please wait, while we are connecting
   you to the sale agent", and is waiting to be connected to the sale
   agent. Then the SN invokes service logic as indicated in the profile.

The execution of this logic selects an appropriate sales agent to
call based on the time of the day. It is 8 P.M. in New York where
the Web user is located, and the New York sales office has closed.
The San Francisco office, however, is still open, and so the SN makes
a call to an agent in that office. Finally, the SN bridges the two
calls and establishes a two-party call between the sales agent and
the Web user.

5.1.3 Web Server-Service Node Interface

Lucent developed the Service Support Transfer Protocol (SSTP) for
communications between the SN and Web Server. SSTP is of a
request/response type running on top of a reliable transport layer,
such as TCP. The Web Server sends a request to the SN to invoke a
service and the SN responds with a message indicating either success
or failure. Note that SSTP engages only the service control function
[1, 2, 3] of the SN.

5.1.3.1 Web Server to Service Node

In this direction, three kinds of messages may be sent: the
Transaction Initiator message, the Data Message, and the End of Data
message.

The latter two messages are needed if the service to be invoked
involves data (such as the case in click-to-fax, click-to-fax-back
and voice-access-to-content). This was so designed to handle the
varying size of data and to ensure that the size of each stream is
within the allowable size of the underlying transport packet data
unit (imposed by some implementations of TCP/IP).

a. Transaction Initiator

This message provides all the necessary information but data for
invoking a service. It includes the following information elements:

+ Transaction ID, which uniquely specifies a service request. The
same transaction ID should be used for all the accompanying data-
related messages, if the service request involves data. One way for
generating unique transaction IDs is to concatenate the information:
date, time, Web Server ID (uniquely assigned for each one connected
to the SN), and transaction sequence number (a cyclic counter
incremented for each service request).

+ Service ID, which specifies the service to be invoked. The service
may be click-to-dial-back, click-to-fax, click-to-fax-back or voice-
access-to-content.

+ Content Provider ID, which uniquely represents the content
provider. This information is the key to accessing the content
provider's service logic and data on the SN.

+ Content Provider Directory Number, which is the telephone or fax
number of the content provider to be called through the PSTN.

+ User Directory Number, which is the telephone or fax number of the
user requesting the service.

+ Billed Party, which specifies the party (either the user or content
provider), to be billed.

In addition, optional parameters may be sent from the Web Server to
the SN. For example, a retry parameter may be sent to specify the
number of times the SN will attempt to complete a service request
upon failure before the transport connection times out.

b. Data Message

This message provides the (encapsulated) user data part of a service
request. For example, in the case of click-to-fax-back such data are
the content to be faxed to the user. Each message is composed of the
transaction ID and a data segment. The transaction ID must be the
same as that of the transaction initiator part first invoking the
service.

c. End of Data Message

This message contains the transaction ID and the end of data
delimiter. The transaction ID is the same as that of the relevant
transaction initiator message.

5.1.3.2 Service Node to Web Server

The SN must respond to a service request from the Web Server. The
response message consists of the information elements:

transaction ID, service type, result, time, and error code.

+ Transaction ID, which is the same as that of the original service
request.

+ Service Type, which is the same as that of the original service
request.

+ Result, which is either success or failure.

+ Time, which indicates the time of the day completing the request.

+ Error Code, which gives the reason for failure. Possible reasons
for failure are content provider telephone (or fax) busy, content
provider telephone (or fax) no answer, user telephone busy, user
refusal to complete, user no answer, nuisance control limit reached,
and content provider telephone (or fax) not in the SN database.

5.1.3.3 Usage Scenarios: Click-to-Fax and Click-to-Fax-Back

For the click-to-fax and click-to-fax-back services, the Lucent
system implemented only the case where the data to be sent as
facsimile reside in the Web server. There are at least three messages
that need to be sent from the Web server to the Service Node for
these services.

The first message is the Transaction Initiator that identifies the
service type as well as a unique Transaction ID. It also includes the
sender/receiver fax number.

The next is one or more messages of the data to be faxed. Each
message carries the same unique Transaction ID as the above.

Last comes the end of message. It consists of the Transaction ID
(again, the same as that of the messages preceding it) and the end of
data delimiter.

Upon receiving these messages, the Service Node, equipped with the
special resource of a fax card, converts the data into the G3 format,
calls the receiver fax, and sends back the result to the Web server
immediately. Note that the receiver fax busy or no answer is
interpreted as failure. Further, while the receiver fax answering the
call is interpreted as success, it does not necessarily mean that the
fax would go through successfully.

5.1.4 Web Server-SMS Interface and SNMP MIB

This interface is responsible for uploading the content provider
profile from the Web Server to the SMS and for managing the
information against any possible corruption. The SN verifies the
Content Provider ID and the Content Provider Directory Number sent by
the Web Server with the content provider profile pre-loaded from the
SMS.

The content provider profile was based on ASN.1 [4] structure and
SNMP [5] was used to set/get the object identifiers in the SMS
database.

Following is an example of the simple MIB available on the SMS.

```
inwebContProviderTable OBJECT-TYPE
        SYNTAX          SEQUENCE OF InwebContProviderEntry
        MAX-ACCESS      not-accessible
        STATUS          current
        DESCRIPTION
                " A table containing Content Provider profiles "
        := { inweb 1}

inwebContProviderEntry OBJECT-TYPE
        SYNTAX          InwebContProviderEntry
        MAX-ACCESS      not-accessible
        STATUS          current
        DESCRIPTION
                " A conceptual row of the inweb. Each row
                        contains profile of one Content Provider"
        INDEX   { inwebSmsNumber }
        := { inwebContProviderTable 1 }

InwebContProviderEntry := SEQUENCE {
        inwebSmsNumber                  Integer32,
        inwebContentProviderId          Integer32,
        inwebContentProviderPhoneNumber Integer32,
        inwebContentProviderFaxNumber   Integer32
        }

inwebSmsNumber OBJECT-TYPE
        SYNTAX          Integer32
        MAX-ACCESS      read-only
        STATUS          current
        DESCRIPTION
                " Serial number of the SMS - used for SNMP indexing "
        := { inwebContProviderEntry 1 }

inwebContentProviderId OBJECT-TYPE
        SYNTAX          Integer32
        MAX-ACCESS      read-create
        STATUS          current
        DESCRIPTION
                " A number that uniquely identifies each Content
    Provider "
        := { inwebContProviderEntry 2 }

inwebContentProviderPhoneNumber OBJECT-TYPE
        SYNTAX          Integer32
        MAX-ACCESS      read-create
        STATUS          current
```

```
         DESCRIPTION
               " Content Provider's Phone Number "
         := { inwebContProviderEntry 3 }

   inwebContentProviderFaxNumber OBJECT-TYPE
         SYNTAX          Integer32
         MAX-ACCESS      read-create
         STATUS          current
         DESCRIPTION
               " Content Provider's Fax Number "
         := { inwebContProviderEntry 4 }
```

5.1.5 Security Considerations

   The Lucent prototype addressed the security issues concerning the
   interface between the Web Server and the SN. Those concerning the
   interface between the Web Server and SMS, which was based in SNMP,
   were handled by the built-in security features of SNMP.

   + Secure Communication Links

   If the Network Operator (PSTN provider) is also the Web Service
   provider, the Web Server and SN/SMS will communicate over a corporate
   intranet. This network is almost always protected by the
   corporation's firewall and so can be deemed secure. This was the case
   handled by the Lucent prototype.

   Nevertheless, if different corporations serve as the Network Operator
   and the Web Service Provider, then it is likely that there may not
   exist a dedicated secure communication link between the Web Server
   and SN/SMS. This raises serious security considerations. One possible
   solution is to use Virtual Private Networks (VPN). VPN features
   support authentication of the calling and called parties and
   encryption of the messages sent over insecure links (such as those on
   the Internet).

   + Non-Repudiation

   All transactions were logged on both the Web Server and the Service
   Node to account for all operations in case of doubt or dispute. The
   log information on the SN may also be used to generate bills.

   + Malicious Requests of Users

   A user may make repeated requests to a content provider directory
   number maliciously. This scenario was handled by setting a Nuisance
   Control Limit (NCL) on either the SN or the Web Server or both. The
   NCL has two parameters: one defining the number of requests from a

user and the other the period over which these requests takes place.

A user may also attempt to request a call from a directory number
other than that of a content provider. This scenario was handled by
verifying the directory number (and the content provider ID) against
the database on the SN containing all the content provider
information. If the directory number (or the content provider ID) was
not in the database, the request would be rejected.

5.2 Siemens Web Call Center

5.2.1 Service Description

The Web Call Center is an Intelligent Network System that accepts
requests from Internet nodes for services to be provided on the PSTN.
As the name suggests, it was designed to support a cluster of
services that, taken together, provide a subset of the features of a
Call Center, with almost all user interactions provided via World
Wide Web requests and responses. See the appendix for a background
description of Call Center Features.

From an Intelligent Network perspective, there are a number of
services that, when combined, provide the Call Center features. The
Call Center features as implemented supported the scenario in which a
customer makes a request to be called back by an agent at a time of
the customer's choosing to discuss an item of interest to him or her.
The agent will be selected based on his or her availability and
expertise in this topic; the agent will be told whom he or she is
calling and the topic of interest, and then the agent will be
connected to the customer.

In addition, the individual services that were deployed to support
this scenario provided support for management of the list of
available agents as well. This involved allowing the agent to "log
into" and "out of" the system and to indicate whether the agent was
then ready to handle calls to the customer. The list of services, as
seen from a user perspective, follows.

The services support:

i) Customer Request service - the customer explores a corporate Web
site, selects a link that offers to request an agent to call the
customer back and then is redirected to the Web Call Center server.
This presents customer with a form asking for name, the telephone
number at which he or she wishes to be called, and the time at which
the call is to be made. Note will also be made of the page to which
the customer was referred to when he or she was redirected. Once the
form has been returned, the customer receives an acknowledgment page

listing the parameters he or she has entered.

ii) Agent Registration/Logon - An agent requests a "login" page on
the Web Call Center server. The service checks whether it has a
record of an agent present at the Internet node from which th call is
made. If not, then the caller will be sent a form allowing him or her
to enter the service identity, the company's agent identifier and
password. On return, the service identity and company agent
identifier will be checked against a list of known identities. If
found, the password will be checked, and if this matches the record
held by the service then a new session record is made of this
identity and the Internet node from which the call has been made.

NB: This is very similar to the Universal Personal Telecommunications
(UPT) service feature "register for incoming calls". It implies that
the identified person has exclusive use of the Internet node from
that point onwards, so messages for them can be directed there.

iii) Agent Ready - an agent who has already logged on can indicate
that he or she is ready by requesting an appropriate "ready" page on
the Web Call Center Server. The service will match the agent by the
Internet node Identifier and Agent Identity passed along with the Web
request against its list of "active" agents. It will mark them as
being ready to handle calls in its list of available agents (with
their pre-defined skill set).

iv) Agent Not Ready - an agent can request an appropriate "ready"
page on the Web Call Center Server to indicate that he or she is
temporarily not ready to handle calls.

v) Agent Logoff - an agent can request an appropriate "Logout" page
on the Web Call Center Server to indicate that he or she is no longer
associated with a particular Internet node. The service will match
the agent by the Internet Node Identifier and Agent Identity passed
along with the Web request against its list of "active" agents. Once
found, the session record for that agent is removed and the caller is
notified of this with an acknowledgment page.

NB: This is very similar to the UPT "unregister" service feature.

vi) Call Center Agent Selection and Notification - When the time
that the customer selected has arrived and an available agent with
the right skills has been selected from the appropriate list, this
service will send a notification to the Internet node associated with
that agent. A dedicated server is assumed to be running on the
agent's machine that, on receiving the notification, triggers the
agent's browser into requesting a "Agent Call In" page from the Web
Call Center Server. Once the agent's machine has made this request,

he or she will be told that there is a customer to call.

NB: This is similar to a "Message Waiting" or "Wake Up Call" service.

Note: As implemented, the agent is led automatically into the
following service (the returned Web page includes an automatic reload
command).

vii) Agent Instruction - a selected agent makes a request of the
"Customer Processing" page on the Web Call Center Server. The
Internet node Identifier and Agent Identity the agent uses will be
matched against a list of agents expected to handle calls, and the
instructions for the calls will be returned to the agent.

NB: This is similar to a "Voice Mail Replay" message service, but in
this case the message is automatically generated; there is no
associated voice mail record feature accessible.

Note: As implemented, the instructions page will include a number of
buttons, allowing the agent to view the page the customer was looking
at when he or she made the request, and to trigger the customer
callback (as described next).

ix) Agent/Customer Telephony Callback - the agent will make a
request of a "dial-back" page on the Web Call Center Server. The
Internet node Identifier and Agent Identity he or she uses will be
matched against a list of agents expected to handle calls, and, when
the appropriate records have been found, the service will make the
telephone call through to the customer and then connect the agent to
this telephone call (using the telephone number registered in the
respective Call Center service record).

5.2.2 Implementation

5.2.2.1 Introduction

The Siemens Web Call Center used an existing IN system and service
logic that supported Call Center features. The scenario it supports
is very similar to the Siemens IN-based Call Center on which it was
based; one of the goals was to minimize changes to the service
offered. It is also virtually identical to the service "Internet
Requested Telephony Dial-back" provided by the Lucent system.

As provided via the Internet, the services involved are mostly the
same as those provided via the PSTN and IN alone. The main
differences lie in the use of the World Wide Web as an interface to
the services rather than a telephone, SSP, and Intelligent
Peripheral. Also, the feature by which a telephone call is made

between the agent and the customer is implemented within the IN
system in a different way; this is the only element in which the PSTN
is involved.

5.2.2.2 Web Call Center Configuration

The general arrangement for the Web Call Center system is shown in
Figure 7. The components that were added to an existing IN system to
deal with the Internet interface are described next.

In addition to the SCP, SSP and SMS that were part of the original
IN-based system, another unit was included to send notification
messages to agents; in the IN system the agents were sent "wake up"
telephone calls when they were required to handle their next
customers' call back. This unit is called the "Internet Intelligent
Peripheral", and its use is described later under "Non-World Wide Web
Interactions".

As there was a need to re-use as many of the existing IN components
unchanged, a Gateway unit to deal with the interface between the
Internet and the SCP was provided. This injected INAP (Intelligent
Network Application Protocol) messages into the SCP, making it think
that it had received an Initial DP trigger from an SSP. It also
intercepted the "Connect To Resource" and "Prompt and Collect" INAP
messages sent from the SCP, acting on these to return the parameters
generated by the Internet users when they filled in the forms that
triggered the service transaction. It also translated the "Play
Announcement" message sent to the Intelligent Peripheral into a form
that it could use. Finally, it passed on the INAP message used by
the SCP to trigger SSP into making the telephone call back.

5.2.2.3 User Interaction

In the IN/PSTN-based system, the services have contact with the
customers and agents via their telephones, SSPs, and Intelligent
Peripherals programmed to play announcements to them and to capture
their responses. These responses are indicated by DTMF tones sent by
pressing keys on the telephones.

In this case, almost all interactions are provided via World Wide Web
requests and responses. The sequence of announcements and responses
for each service are "collapsed" into individual form filling
transactions, and the requests are not limited to digits (or "star"
and "hash"). The implications of the use of forms on service
operation are covered in more detail later (under HTTP/IN Service
mapping).

5.2.2.4 Service/Caller Identifiers

   When provided via the IN/PSTN-based system, the services are passed
   the Calling Line Identity (CLI) of the caller and the number the
   caller dials (the DN). The CLI value is used extensively to identify
   the caller and (in the case of the agent) to index into service data
   tables to decide what to do next. While an equivalent value to the
   DN is passed to the Web-based transactions as the requested Universal
   Resource Locator (URL), the CLI cannot be given reliably. The nearest
   equivalent caller identifier is the IP Address of the customer or
   agent's machine. However, the use of HTTP proxies means that this
   "original" Internet node Address may not be available; if a proxy is
   used then its IP Address will be associated with the request.

   In providing these Call Center features the customer only has one
   Web-based transaction; that of providing the initial request for a
   PSTN telephone callback. To do so he or she will have to fill in a
   form so as to specify not only the time to be called back, but also
   the telephone number to be reached. These values can be used if
   needed to identify the customer, and so the problem of originating
   Internet Node ambiguity is not relevant.

   With the agents, however, there are sequences of coupled
   transactions, and the particular sequence must be identified. There
   will be a number of such transactions being carried out at once, and
   there needs to be some identifier to show which agent is being
   handled in each case.

   Such an identifier is not part of a sequence of basic Web
   transactions. In a Web transaction, the HTTP Client/Web Browser makes
   a request, and the HTTP Server will respond to this, normally
   including some content in its reply message that will be processed by
   the browser, after which it closes the TCP connection. That's the end
   of the transaction; the HTTP client and server cannot normally
   maintain state information beyond this point. Any sequence is reduced
   to a set of unrelated transactions.

   A result of this simple pattern is that any state information
   reflecting longer or more complex interactions must be stored (at
   least partially) in the client system. One approach is the use of
   cookies [6]. These can be set by HTTP servers as part of their
   response to a request, and will be sent back with all subsequent
   requests for appropriate URLs as extra HTTP headers. These cookies
   allow the HTTP server to identify the client in the following
   requests, so that it can continue an extended session with the
   client.

Cookies are used in providing the Internet Call Center. Persistent
cookies are installed into the Web Browser on machines that are to be
used by call center agents as a service management (pre-service)
task. The cookie value is unique to the machine and is used to index
into a list of machine IP addresses that is stored as part of the
service data.

Also, a session cookie is stored onto the agent's machine when the
agent registers, and is cleared when he or she de-registers. This is
used to identify the agent and so the IP address of the node with
which the agent is associated (and from which the agent's subsequent
requests should originate). The services that interact with Call
Center agents use the agent session cookie value as an identifier; in
principle this is unnecessary but it does simplify the session data
lookup procedure. The rest of the services use the persistent machine
identifier in place of the CLI, indexing into their service data
using it. Both cookies are sent with each agent request; if they are
not present, then the request is redirected to other services (for
example to the agent Logon service).

5.2.2.5 Mapping from HTTP Transactions to IN-Based Service Features

All of the client-initiated services require user interaction. With
the IN/PSTN-based system, the majority of the services are typified
by the callers being connected to an announcement unit that plays
them a list of choices and captures their selection. The caller can
pre-dial the digits needed; in this case the prompts are not needed
and are not made.

The pattern of operation is somewhat different in the Internet case,
as the initial HTTP request returns a response, after which the Web
transaction has ended. Where that initial response returns a form to
be filled in by the caller, subsequently submitting the form
initiates a new HTTP transaction. This is all part of one instance
of service, however. The service consists of two request/response
pairs in tandem.

Although it is possible to design a service to handle this pair of
Web transactions as a single unit, it may be better to reconfigure
it. The design of a service that deals with two Web exchanges as a
single extended transaction is quite complex. It must maintain state
across the pair of Web exchanges, and it has to handle a number of
failure cases including dealing with time-outs and "out of time"
submission of forms. The alternative is to split the service into two
sub-features. The first of these reflects the initial request and
delivery of the form by return, with the second one dealing with
processing of the submitted form and returning any confirmation by
reply.

The services offered don't all require form-filling, and so can be
treated as a single IN feature. There are two cases where forms are
required. The first of these is the Customer Request service, while
the other one is the "Agent Registration" service. In both cases the
initial Web transaction (by which the form is requested and returned
to the client) need not involve specific service logic processing;
the initial delivery of the form to a customer or agent can be
handled by a "normal" Web Server. In both cases the service logic is
only triggered when the form is submitted; this means that, again,
each of the services can be treated as a single IN feature.

The IN service logic that deals with these requests has a general
pattern of action. An HTTP request is received, and this triggers the
IN service logic into action. The service logic "sees" this as an
Initial DP message and starts its processing as if it had been sent
from an SSF. The SCF uses what appears to it to be an Intelligent
Peripheral to collect the parameters of the request, and then to send
back final announcements to the requesting entity.

The main difference, from the perspective of the IN service logic
running on the SCF, is that the service does not need to instruct the
SSF to make a temporary connection to the Intelligent Peripheral. It
is as if this connection had already been made. Similarly, there is
no need to close the service transaction by sending an explicit
"Continue Execution" message to the SSF.

The sequence of "prompt/collect" instructions used to collect service
parameters from a caller in an IN service maps quite well to a
sequence of requests to extract a data value from the HTTP request,
based on a tag. This is a fairly standard feature of Web Server CGI
or Servlet processing. Using this mapping minimizes the changes to
the service design, in that the service logic "sees" an Intelligent
Peripheral to which it sends normal "Request Report Prompt & Collect"
messages, and from which it receives data values in response.

All services have to fit in with the underlying HTTP interaction
pattern, and so will be expected to send a final "Announce"
instruction to the Intelligent Peripheral at the end of the service;
this is done in many IN services anyway and in all of the service
features described here. These announcements form the content
returned to the Web Client.

5.2.2.6 Non-World Wide Web Interactions

   There are two exceptions to the sole use of the World Wide Web for
   interaction. The first one occurs in the "Message Waiting"/"Wake Up
   Call" service by which the selected agent is informed of a callback
   request. World Wide Web transactions are very simple; the client
   browser makes a request for content associated with a particular HTTP
   URL, and the server sends a response, marking the end of the
   transaction. The server cannot make a spontaneous association with a
   client; it must be initiated by the client request.

   While it would be possible for the server to defer closing an earlier
   transaction (by not sending back all of the content specified and
   leaving the TCP connection open) it was decided that an alternative
   scheme would be more convenient. The "wake up call" was arranged by
   an "Internet Intelligent Peripheral" sending a request to a daemon
   process running on the selected agent's machine, using the Finger
   protocol [7]. The daemon sent back a standard response, but in
   addition the Web Browser on the agent's machine was triggered into
   making a further HTTP request of the server. In this way the "Agent
   Instruction" transaction is started automatically, while still
   allowing it to use a normal HTTP request/response pattern.

   The second exception occurs in the final "Agent/Customer Telephony
   Callback" service. While this transaction is initiated by the agent
   selecting a link on the "call instructions page" returned to them,
   and includes a "confirmation" page being sent back to them in an HTTP
   response, the purpose of this service is to make a telephone
   connection via the PSTN between the agent's telephone and the
   customer's telephone. It is the only service element that involves
   the PSTN directly. From an IN/PSTN perspective, the resulting
   telephone connection is different from that provided in the scheme
   using the IN and PSTN alone. In this case, a PSTN call is made out to
   the agent's telephone, another call is made out to the customer's
   telephone, and these calls are bridged. This differs from the earlier
   scheme, in which the agent originated a call to the voice mail replay
   system, and this call was redirected to a new destination (the
   customer's telephone). As this feature differs in purpose from the
   other services, and it requires a different implementation within the
   IN and PSTN system, it was organized as a separate service in this
   case.

5.2.2.7 Security Considerations

   In the case of this system, assumptions were made that the interface
   presented to requesting agents and customers was provided via a fire
   wall to deal with most attacks on the IN components. The interface
   appeared as a Web Server, and there was no direct access to the HTTP
   documents served, nor to the servlets providing the service logic.

   The Callback service was deemed to have simpler security requirements
   than other IN services as it was akin to a free phone "1-800" service
   access number; the agents work for the service subscriber and are not
   charged directly. Similarly, the requesting customer is not charged
   for his or her request, nor for the resulting call back. Service
   subscribers would be willing to pay the costs of telephone calls
   generated as a result of this cluster of services, and the costs of
   running the agent services could be charged directly to them. As such
   the authorization for service is defined by the contract between the
   service subscriber and the service provider.

   Authentication of agents was seen as a problem. As an interim
   measure, cookies were used, but this scheme delivers the cookie data
   as a plain text item (a header of the Web request). Secure Socket
   Layer connections were required for communication with the agent
   services, and this had an impact on the performance of the IN system.

5.2.3 Derived Requirements/Lessons

   Security is seen as a major issue. A firewall was used to control
   access to the IN Components. Similarly, SSL was used for
   communication with the Agents, so as to protect the cookie values
   that they were sending with their requests.

   For other services, it is likely that the entity from which requests
   appear to originate will be charged for the service to be rendered.
   This has implications in terms of authentication and authorization of
   service provision at the time of the request. It is necessary for the
   service to be authorized in such a way that non-repudiation is
   ensured; this is likely to mean that a certificate of identity be
   provided from the person making the request, and that this can be
   tied in with a financial account that that person has with the
   service provider. The certificate can then be stored as part of the
   billing record. While the process of electronic commerce is outside
   of the scope of this work, the mechanism by which a request for
   confirmation of identity is passed out to the requesting user and is
   delivered back to the service logic must be considered.

When changing from a "pure" IN/PSTN system to one supporting requests
via the Internet, the differences in the way that clients interacted
with the services meant that the service logic had to be redesigned.
It was realized that maintaining the state of a service during its
processing was going to be a problem; this problem was side-stepped
by re-engineering the services as form processors, allowing them to
deal with fully specified requests as a single (Web) transaction. In
addition, a "normal" Web Server was used to deliver the forms to the
users. This is a change from the IN system, where the equivalent of
the form (the prompts) were sent in sequence as part of the same
service process.

The Call Center features provided suited this change. However, this
may not be the case for other IN services. It is quite common for
services to be designed such that the user is prompted for a
response, and the service continues dependent on this response. The
Web form presents all of the options at once, so this kind of variant
prompt/collect sequence is not possible. From this, it is difficult
to see how an IN service could be reused without some degree of
modification.

An intermediate "gateway" system was provided to "cocoon" the service
logic as far as possible from the details of the components with
which it was working. Where needed, this unit translated calls from
the service logic into commands that operated with the Internet (and
the Web Server that acted as the interface). Our experience was that
an SCP could be "spoofed" into thinking that it was operating with
other IN components in the normal way. Within the limits of the
service used, this proved simpler than was originally expected.

Selecting this simple approach still allows a considerable range of
services to be provided while maintaining any investment in existing
IN systems. Modification of existing IN service logic was also
easier than feared. All of the services examined provided
announcements at the end of the service transaction, and this could
be used to trigger a Web response to be sent back to the requesting
Internet user. The changes to the Call Center service logic turned
out to be minor; it took as long to analyze the service and see how
it could be arranged as a sequence of "form processing" transactions
as it did to make the changes to the service logic.

In the Siemens Web Call Center, the "Internet Intelligent Peripheral"
with which the service logic communicated was running as a separate
program on the same node. Where more complex behavior is required of
it (such as conversion of text to speech data and interface with the
PSTN) then it would almost certainly be on a separate node. If data
is transferred from the Internet in such a scheme, any intermediate
gateway would be involved in relaying the data to this node.

6. Alternative Solutions

6.1 The AT&T System

   AT&T developed a framework for controlling voice and voice-band data
   (e.g., fax) and for providing PINT services. Key to the framework is
   CallBroker, a logical entity that acts on behalf of a user to set up
   sessions and make requests for PSTN resources. The sessions typically
   include initiation of calls between two or more end points specified
   by the user. In addition to its interactions with the PSTN for call
   setup, the CallBroker is responsible for other functions, when
   necessary, such as authentication and usage recording.

   This section briefly discusses the protocol at the two interfaces
   that need to be defined and the corresponding APIs to provide the
   above services. The two interfaces are (1) the one between the
   CallBroker (or Web Server) and the Service Control Function in the
   Service Node in the PSTN and (2) the one between the IP client and
   the CallBroker. The latter interface, in particular, will enable
   service providers to extend the architecture defined here to serve as
   a platform for other advanced/value-added services (to be identified
   later). In addition, the view taken here is that the IP client is
   more general, and implements a protocol for communication with the
   CallBroker that allows full two-way communications. For example, this
   is required for the cases where a called party hangs up and an
   indication may be necessary to be given to the IP Client about this
   status/progress. This is also necessary when conferencing to give an
   indication/status of various parties joining the call.

6.1.1 High Level Architecture

   A high level architecture depicting various logical entities and the
   Interfaces among these logical Entities and the IP Client is shown in
   Figure 12.

```
                               _____
                              /
              1           ____     / 2     ____
    /|_____  |    | |_____  |    |       PSTN
   / |_____  |____| \           |____|
                          Call    \      /  SCF\
                          Broker    \    /   SN  \
                                     _____
                                    /           \
                                   /             \
                                  /               \
                                 __                __
                                /\                 /\

                          Calling          Participant
                          Party           (Called Party)
```
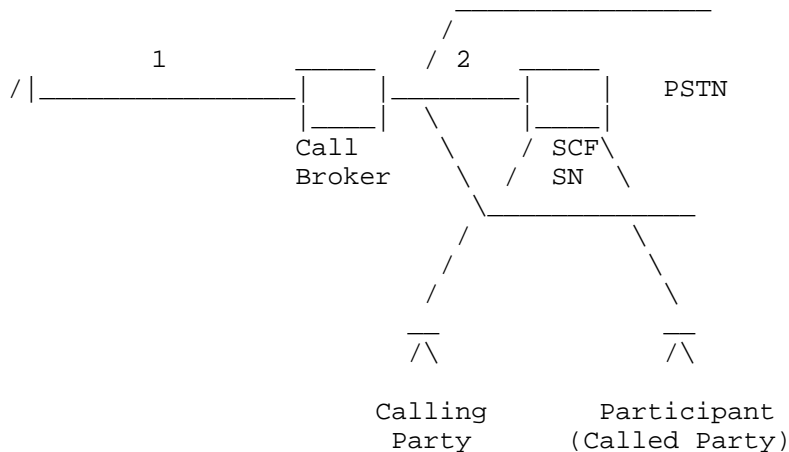
                Figure 12:   The CallBroker Architecture

   The CallBroker, in addition to the initiation and control of calls on
   behalf of the user, performs additional functions. These functions
   include authenticating the IP Client, usage recording, and management
   of the session for the IP Client for the telephony call. The notion
   of the session requires that a client state machine be maintained in
   the CallBroker. This also helps in notifying the IP Client about the
   status/progress of the requests generated from the IP Client.

   From the perspective of the IP Client, the logical entities needed
   for the above functions are within the CallBroker and are as shown in
   Figure 13 below. These correspond to the functions already
   discussed: Usage Recording Function, Session Management Function,
   Voice Bridge, and the Authentication Function. The fact that some of
   these functions may be physically separate from the CallBroker (such
   as the Voice Bridge being in the PSTN) is not inconsistent with the
   general view adopted here. Thus, the CallBroker Model mediates
   requests for network services and enables us to define various value
   added services in the future.

```
lllllllllllllllll
l               l
l Call Broker  l                      Authentication
l  Server      l                      Function
l  _____      l        Interface 2a    _____
l |      |x x xlx x x x x x x x    |    |
l |_____|x     l                      |____|
l        x    x l
l         x    xl   Interface 2b
lSession State lx
l   Mnmgt. x  l  x              Usage Recording
l  Function    l     x              Function
l  _____    x l       x              _____
l |      |    l              x  x x  |    |
l |_____|     xl                      |____|
lllllllllllllllll
               x
                x Interface 2c
                x
             _____
            |        |
            |_____|

             Bridge
```
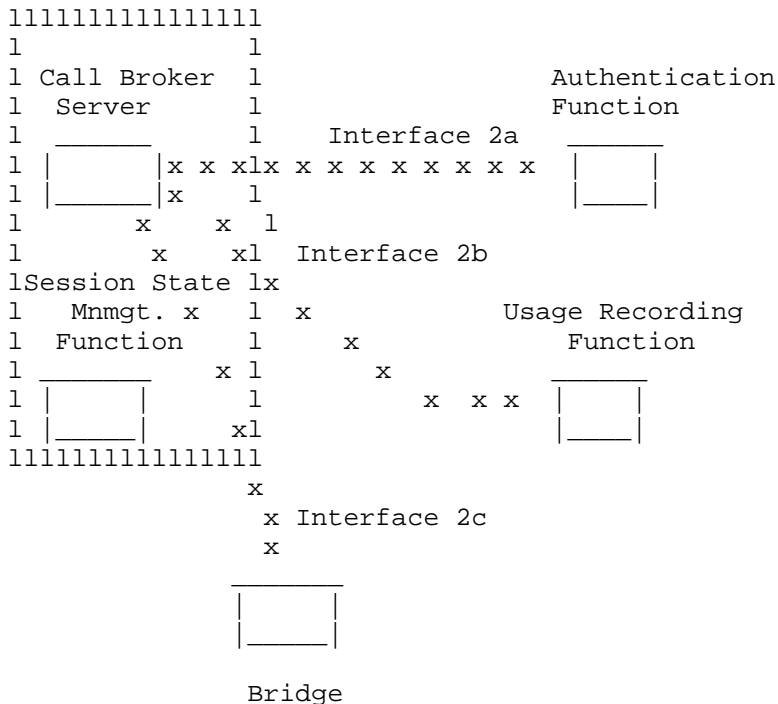
        Figure 13: Functional Entities in the Call Broker

   Various interfaces (i.e., 2a, 2b, 2c in Figure 13) between different
   functional entities in the CallBroker may also be standardized. The
   Session State Management Function may be physically realized as part
   of the CallBroker Server.

6.1.2 IP Client to CallBroker Interface

   Communication on the IP Client to CallBroker Interface (Interface 1
   in Figure 12) is a simple ASCII based protocol running directly on
   TCP. The messages on this interface are primarily requests from the
   client to the CallBroker, responses from the CallBroker to the IP
   client responding to the requests and unsolicited events from the
   CallBroker to the IP client. Since the communication is not strictly
   transaction oriented, traditional encapsulation protocols like HTTP
   cannot be used. There has been some ongoing work attempting to use
   multiple concurrent HTTP POST requests to support event delivery but,
   without too much difficulty, the ASCII protocol specified here can
   easily be mapped to the POST payload of the HTTP protocol.

6.1.3 Protocol

   Basic Format

   The basic format of the protocol is as follows:

   [header]<<LF>
   <<LF>
   [body]<<LF>
   <<LF>
   <<LF>

   The header and body of the protocol are separated by 2 line feed
   characters. The format of the header and the body is described
   below. Line feed characters in the header or body will be escaped
   using simple URL encoding.

   Header

   [session-id | 0]<<LF>
   [message-id]<<LF>
   [version-info]<<LF>

   All CallBroker transactions are identified by sessions. A session
   does not necessarily correspond one-to-one to a TCP session. If the
   IP client is attempting to initiate a new session with the CallBroker
   the session-id field is populated with '0' to indicate session
   creation request. Every session request needs to be accompanied by
   sufficient information regarding authentication for the CallBroker to
   create the session.

   Message-id represents the operation of the message.

   Version-info contains optional version information of the protocol.
   This is to aid possible version mismatch detection and graceful error
   recovery.

   Body

   The body of the protocol messages consists of name value pairs. These
   name-value pairs are interpreted with reference to the message-id
   which signifies the operation to be performed by the CallBroker.

6.1.4 APIs Exposed to the IP Client

   The APIs of the CallBroker exposed to the IP client are distinct and
   different from the APIs that the CallBroker uses from the different
   supporting subsystems including the authentication subsystem and the
   usage recording subsystem. The IP client APIs enable clients to
   effectively control voice conferencing.

6.1.5 Voice-Bridge Control API

   The Voice Bridge Control API is used by CallBroker applications to
   access voice bridging functionality. The API distinguishes between
   sessions and calls. Calls represent actual voice calls placed from/to
   the voice bridge. These calls can be grouped together in sessions.
   All the calls that belong to a session are bridged. Calls have a
   significance outside the scope of sessions. Every call can be
   associated with multiple sessions with different weights at the same
   time. The advantage of this approach is the ability to support
   concepts like whispering in a conference call. Calls can also be
   dropped from a conference session and bridged together in a new
   session to give the notion of a sub-conference. These calls can later
   be re-added to the main conference session.

6.2 Simple Computer Telephony Protocol

6.2.1 Overview

   The Simple Computer Telephony Protocol (SCTP) is a third party call
   control protocol and as such does not comply with the PINT charter.
   SCTP is described in this section to show how PINT services could be
   implemented using SCTP, and where SCTP fits into the PINT
   architecture.

   In addition to third party call control, SCTP also provides
   subscriber (i.e., user) feature management (e.g., allows a user to
   set do not disturb, call forwarding parameters), and subscriber
   monitoring of terminal, line and address status. SCTP is strictly
   client/server-based. It has no provisions for peer to peer
   communications. SCTP runs as a TCP application protocol. It is
   ASCII-based and uses sockets. The SCTP Server is usually connected to
   a switch via a CTI (Computer-Telephony Integration) connection.
   Because of this, feature interactions are limited to those within the
   context of a single call, and not between PSTN services. The SCTP
   Server within a PINT Gateway could also be connected to an SN, or an
   SCP. See figures below. SCTP does NOT carry media.

6.2.2 How SCTP Fits in with the Reference PINT Services

   SCTP Client as Part of a Web Server

```
  +------+    +--------+         +--------+    +------+
  |      |    |        | SCTP    |        |    |      |
  |      |----|        |-------  |        |----|      |
  |      |    |        |         |        |    |      |
  +------+    +--------+         +--------+    +------+
  User's PC   Web Server/        PINT Gateway  SN/SCP/Switch
              CGI
```
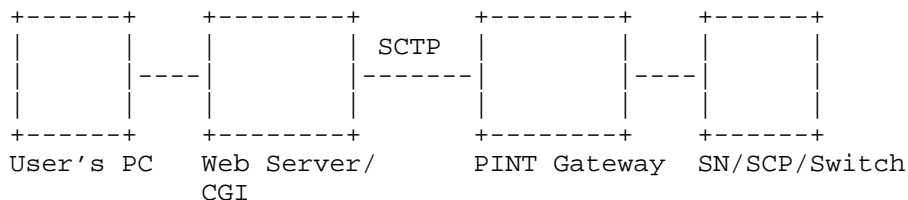
              Figure 14: SCTP Client as Part of a Web Server

   In this architecture, the SCTP Client is embedded in the Web Server.
   It is there for the specific purpose of initiating calls to the PSTN
   based on user requests. The SCTP Server is within the PINT Gateway.
   We go through the classic PINT examples:

   Click-to-dial-back: The SCTP Client issues an SCTP MakeCall to the
   SCTP Server with the calling number supplied by Web page, and called
   number supplied by the user.

   Click-to-fax-back: SCTP Client issues an SCTP MakeCall to the SCTP
   Server with called number set to user's fax machine, and calling
   number set to Web Server's fax machine, and treatment set to the URI
   for the file to be faxed. The SCTP Server takes the file and feeds
   it into the call just as a fax machine would.

   Click-to-fax: SCTP Client issues an SCTP MakeCall with calling number
   set to user's fax machine, and called number set to Web Server's fax
   machine. How the file is supplied to the user's fax machine is
   outside the scope of SCTP.

   Voice-access-to-content: SCTP Client issues an SCTP MakeCall with
   called number set to user's telephone number, and calling number set
   to Web Server and treatment set to a URI for the file of the
   particular Web page to be read to the called number. The SCTP Server
   takes care of the file to voice conversion and this is fed into the
   call as if it were voice.

   In all of the above cases, the SCTP Client can generate a variety of
   different Web pages to send to the Web Server via CGI (Common Gateway
   Interface). The content of these pages is based on the call
   completion status of the CallMake SCTP action.

   SCTP Client Running on the User's PC


```
                                     +------+
                    HTML             |      |  INTERNET
        +-----+    /-------------|      |
        |     |---/                 +------+
        |     |                     Web Server
        |     |---\
        +-----+    \
        User's PC   \ SCTP       +------+       +------+
                     \-----------|      |-------|      |  PSTN
                                 |      |       |      |
                                 +------+       +------+
                                 PINT Gateway   SN/SCP/Switch
```
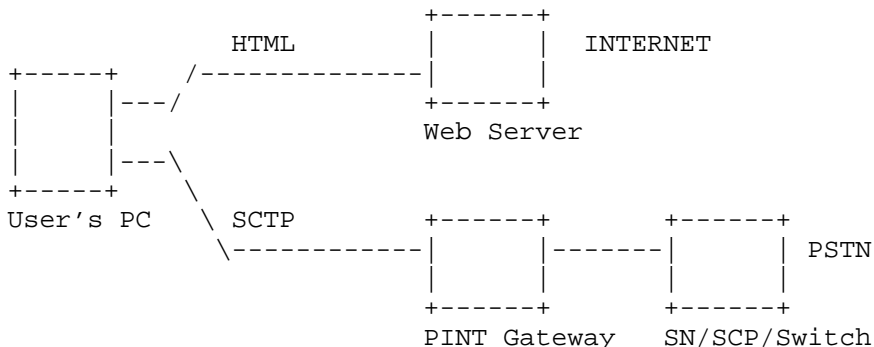

            Figure 15: SCTP Client Running on the User's PC

   In this architecture, the user has an SCTP Client co-located with it.
   If the user is using the telephone line for connection to a Web
   Server and there is an incoming call, then the SCTP Server in the
   PINT Gateway will post this event to the SCTP Client. A window will
   pop up on the user's screen with options available to the user for
   handling of the incoming call. The user can choose to take the call,
   send it to voice mail, or send it to another number.

   For the Fax back service, for example, if the user had a separate fax
   machine from his or her PC, then the SCTP Server would tell the SCTP
   Client there is an incoming fax. The user would end or suspend his or
   her Internet connection, the fax would come in, and the user could
   then resume the Internet connection.

7. Session Initiation Protocol--An Emerging Standard

7.1 Overview

   SIP, the Session Initiation Protocol, is a simple signaling protocol
   for Internet conferencing and telephony. It is currently under
   development within the IETF MMUSIC (Multiparty Multimedia Session
   Control) Working Group.

   SIP provides the necessary mechanisms to support the following
   services:

   - call forwarding, including the equivalent of 700-, 800- and 900-
     type calls;
   - call-forwarding no answer;

- call-forwarding busy;
- call-forwarding unconditional;
- other address-translation services;
- callee and calling "numbers" delivery, where the numbers can be of
  any (preferably unique) naming scheme;
- personal mobility, i.e., the ability to reach a called party under
  a single, location-independent address, even when the user changes
  terminals;
- terminal-type negotiation and selection: a caller can be given a
  choice of how to reach a party, e.g., via Internet telephony,
  mobile, phone, and an answering service;
- caller and callee authentication;
- blind and supervised call transfer;
- user location; and
- invitation to multicast conferences.

Extensions of SIP to allow third-party signaling (e.g., for click-
to-dial-back services, fully meshed conferences and connections to
Multipoint Control Units (MCUs), as well as mixed modes and the
transition between those) have been specified.

SIP addresses (URLs) can be embedded in Web pages. SIP is
addressing-neutral, with addresses expressed as URLs of various types
such as SIP, H.323 or telephone (E.164). A purely representational
example of a SIP URL might be sip:+12125551212@foo.example.com, where
foo.example.com is the host serving as a gateway into the PSTN.

SIP is independent of the packet layer and only requires an
unreliable datagram service, as it provides its own reliability
mechanism. While SIP typically is used over UDP or TCP, it could,
without technical changes, be run over IPX, or carrier pigeons, ATM
AAL5 or X.25, in rough order of desirability.

SIP can set up calls "out-of-band". For example, while the SIP
protocol exchanges use IP, plus UDP or TCP, the actual data transport
can take place via the PSTN. This feature makes it possible to use
SIP to control a PBX or send requests to a Service Control Point. The
PINT services make use of this flexibility.

7.2 SIP Protocol

SIP is a textual client-server protocol, similar in syntax to HTTP
and RTSP. Requests consist of a method (INVITE, BYE, ACK, or
REGISTER), a list of parameter-value pairs describing the request and
an optional request body. Parameters include the origin and
destination of the call and a unique call identifier. They may
indicate the caller's organization as well as the call's subject and
priority. The request body contains a description of the call to be

established or the conference to be joined. The description format is
not prescribed by SIP; SDP is one possibility being standardized
within the IETF. For the purposes of providing PINT services, an
additional phone number address format is to be added to SDP.

Responses indicate whether a request is still being processed, was
successful, can possibly be satisfied by another node or failed. When
a call is redirected, the response indicates the name of the node to
be tried. Unsuccessful calls may also return a better time to try
again.

In a typical successful call, the caller sends an INVITE request to
the callee. The callee accepts the call by returning a response code
to the callee, which then confirms the receipt of that acceptance
with an ACK request. Either side can terminate the call by sending a
BYE request.

Requests can be authenticated using standard HTTP password and
challenge-response mechanisms. Requests and responses may also be
signed and encrypted.

7.3 SIP entities

SIP distinguishes three kinds of entities:

User agents receive and initiate calls and may forward the call.

A proxy server is an intermediary program that acts as both a server
and a client for the purpose of making requests on behalf of other
clients. Requests are serviced internally or by passing them on,
possibly after translation, to other servers. A proxy must interpret,
and, if necessary, rewrite a request message before forwarding it. A
proxy server may, for example, locate a user and then attempt one or
more possible network addresses.

Redirect server accepts a SIP request, maps the address into zero or
more new addresses and returns these addresses to the client. Unlike
a proxy server, it does not initiate its own SIP request. Unlike a
user agent server, it does not accept calls.

Proxy and redirect servers may make use of location servers that
determine the current likely location of the callee.

A PSTN gateway initiates phone calls between two parties. This may be
a server that sends requests to an SCP in an IN environment or it may
be a CTI-controlled PBX.

A SIP call may traverse one or more proxy servers.

   The servers that control a PBX or an SCP act as user agents. A Web
   server may also act as a SIP user agent.

7.4 Providing Call Control Functionality

   The SIP for PINT specification provides details on how to use SIP to
   initiate phone calls between two PSTN end points. (SIP can also
   initiate calls between Internet end points and between an Internet
   and PSTN end point, but this is beyond the scope of this document.)

   It should be noted that the SIP client for initiating such phone
   calls can be either at the user's location (his/her workstation) or
   can be a Web server that calls up a SIP client via a CGI program.
   There is no difference in operation or functionality, except that the
   owner of the Web server may be legally responsible for the calls
   made.

   A SIP client needs to convey two addresses to the PSTN gateway: the
   party making the call and the party to be called. (The party to be
   billed also needs to be identified; this can either be done by a SIP
   header or by having the server look up the appropriate party based on
   the two parties. This aspect is for further study.)

   Described below are three ways these addresses can be conveyed in
   SIP. In the example, the address of party A is +1-212-555-1234 and
   that of party B is +1-415-555-1200. (The URL types in this and other
   examples are representational; they may but do not have to exist.)

   (1) The two PSTN addresses are contained in the To header (and
   request-URI) and an Also header. For example:

     INVITE sip:+1-212-555-1234@pbx.example.com SIP/2.0
     To: phone:1-212-555-1234
     From: sip:j.doe@example.com
     Content-type: application/sdp
     Call-ID: 19970721T135107.25.181@foo.bar.com
     Also: phone:+1-415-555-1200

     v=0
     o=user1 53655765 2353687637 IN IP4 128.3.4.5
     c=PSTN E.164 +1-415-555-1200
     t=0 0
     m=audio 0 RTP/AVP 0

   In that case, the gateway first connects to party A and then party B,
   but without waiting for A to accept the call before calling B.

(2) Parties A and B are indicated by separate invitations. This
allows the gateway to make sure that party A is indeed available
before calling party B. After calling party A, the gateway could
play an announcement indicating that the call is being connected
using, for example, RTSP with appropriate Conference header
indicating the call.

```
INVITE sip:+1-212-555-1234@pbx.example.com SIP/2.0
To: phone:1-212-555-1234
From: sip:j.doe@example.com
Content-type: application/sdp
Call-ID: 19970721T135107.25.181@foo.bar.com
...
INVITE sip:+1-415-555-1200@pbx.example.com SIP/2.0
To: phone:+1-415-555-1200
From: sip:j.doe@example.com
Content-type: application/sdp
Call-ID: 19970721T135107.25.181@foo.bar.com
...
```

(3) The two PSTN addresses are conveyed in the To header of the SIP
request and the address in the SDP media description. Thus, a request
may look as follows:

```
INVITE sip:+1-212-555-1234@pbx.example.com SIP/2.0
To: phone:1-212-555-1234
From: sip:j.doe@example.com
Content-type: application/sdp
Call-ID: 19970721T135107.25.181@foo.bar.com

v=0
o=user1 53655765 2353687637 IN IP4 128.3.4.5
c=PSTN E.164 +1-415-555-1200
t=0 0
m=audio 0 RTP/AVP 0
```

Here, pbx.example.com is the name of the PSTN gateway; the call will
be established between 1-212-555-1234 and +1-415-555-1200.

Users can be added to an existing call by method (1) or (2).

8. Overall Security Considerations

Inter-networking of the Internet and PSTN necessitates the
introduction of new interfaces (e.g., the A, B and E interfaces in
Figure 6). To ensure that their use does not put the networks, in
particular the PSTN, at additional security risk, these interfaces
need to be designed with proper security considerations. Sections

5.1.5 and 5.2.2.7 describe how two of the pre-PINT implementations,
the Lucent and Siemens systems, handle the security aspect,
respectively.

Worth noting are the security requirements suggested by pre-PINT
experiences. They are:

+Peer entity authentication to allow a communicating entity to prove
its identity to another in the network (e.g., the requesting IP-host
to the PINT gateway, and the PINT gateway to the PSTN node providing
the service control function).

+Authorization and access control to verify if a network entity
(e.g., the requesting IP-host) is allowed to use a network resource
(e.g., requesting services from the PINT gateway).

+Non-repudiation to account for all operations in case of doubt or
dispute.

+Confidentiality to avoid disclosure of information (e.g., the end
user profile information and data) without the permission of its
owner.

In the course of the PINT interface development, additional
requirements are likely to arise. It is imperative that the resultant
interfaces include specific means to meet all the security
requirements.

9. Conclusion

   This document has provided the information relevant to the
   development of inter-networking interfaces between the PSTN and
   Internet for supporting PINT services. Specifically, it addressed
   technologies, architectures, and several existing pre-PINT
   implementations of the arrangements through which Internet
   applications can request and enrich PSTN telecommunications services.
   One key observation is that the pre-PINT implementations, being
   developed independently, do not inter-operate. It is a task of the
   PINT Working Group to define the inter-networking interfaces that
   will support inter-operation of the future implementations of PINT
   services.

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11. Appendix

11.1 PSTN/IN 101

11.1.1 Public Switched Telephone Network

   What is normally considered as "the Telephone Network" consists of a
   set of interconnected networks. Potentially, each of these networks
   could be owned by a different Network Operator. The official name for
   such a network is Public Switched Telecommunications Network (PSTN).
   A simple PSTN consists of a set of Switches (called Central Offices
   or Telephone Exchanges) with links interconnecting them to make up
   the network, along with a set of access connections by which
   terminals are attached. The PSTN is used to deliver calls between
   terminals connected to itself or to other PSTNs with which it is
   interconnected. Calls on the PSTN are circuit switched; that is, a
   bi-directional connection is made between the calling and called
   terminals for the duration of the call. In PSTNs the connection is
   usually carried through the network in digital format occupying a
   fixed bandwidth; this is usually 56 or 64 Kbps. The overall
   configuration of the PSTN is shown in Figure 16.

```
  /--\
 ()/\()__
  /__\   \      .................................
        \       !             !                   !           /--\
   __     \   [-!-]         [-!-]                  !          ()/\()
   \ \      \__[CO ]========[CO ]==\\              !        ___/__\
  [Fax]_____[---]         [---]  \\         [-!-]      /   __
                                     \\======[CO ]____/     \ \
                                             [---]_____[Fax]
   Key: ___    Access Lines
        ===    Trunk Links (inter-CO user data links)
        ...    Inter-CO signaling network links
```
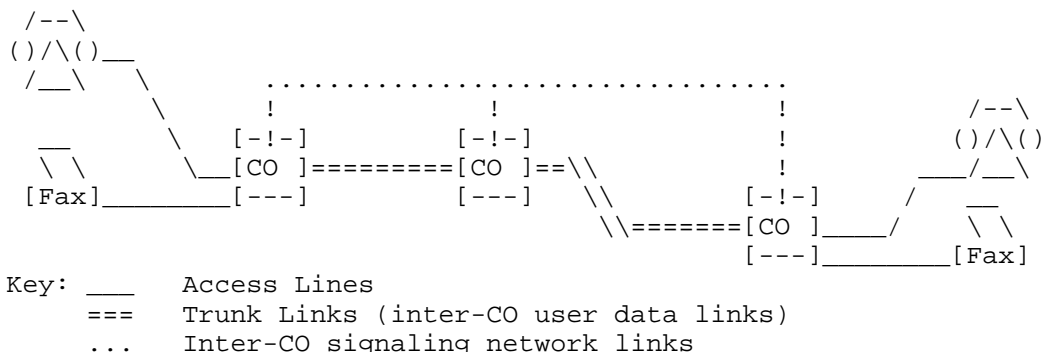
                            Figure 16

   Messages are sent between the Switches to make and dissolve
   connections through the network on demand and to indicate the status
   of terminals involved in a call; these "signaling" messages are
   carried over a separate (resilient) data network dedicated to this
   purpose. This signaling network is also known as the Common Channel
   Signaling (CCS) or Signaling System Number 7 (or SS7) network after
   the names of the signaling protocol suite used.

   As yet, the majority of access connections to a PSTN carry analogue
   signals, with simple (analogue) telephones or Facsimile machines as
   terminals. Call requests are indicated to the Central Office to which

a telephone is connected either by a sequence of pulses or tone pairs
being sent. Notifications on the status of the request are sent back
to the telephone in the form of tones. Indication from a Central
Office that a call is being offered to a telephone is arranged by
sending an alternating voltage down the access connection which in
turn causes the ringer in the telephone to sound. These access lines
have a unique address associated with them and can support a single
call.

However, with analogue or digital multi-line connections, or
Integrated Service Digital Network (ISDN) Basic or Primary Rate
Interfaces (BRI or PRI), several concurrent calls are possible and a
set of addresses are associated with them. The new ISDN access
connections are designed so that data exchanged with the network is
in multiplexed digital form, and there is an individual channel for
each of the potential connections, together with a separate channel
dedicated to sending and receiving call request and call alert data
as well as carrying packet switched user data. These call request and
call alert messages act as the equivalent of the pulses or tones that
are sent when dialing, and the ringing signal that is sent to a
telephone when a call is being made to it.

The operation of the call request is fairly simple in most cases and
is shown in Figure 17.

```
  /--\
 ()  ()
  --____
 /++\   \        ..................................            /--\
/----\   \      ^              v                    !          ()  ()
  A      \   [-!-]         [-!-]                     !            --
          \->[CO ]========[CO ]==\\                  v        ->-/  \
            [---]          [---]   \\        [-!-]        /  /----\
                                    \\======[CO ]____/      B
                                            [---]
  Key: ___     Access Lines
       ===     Trunk Links (inter-CO user data links)
       ...     Inter-CO signaling network links
       CO      Central Office (Telephone Exchange)
```
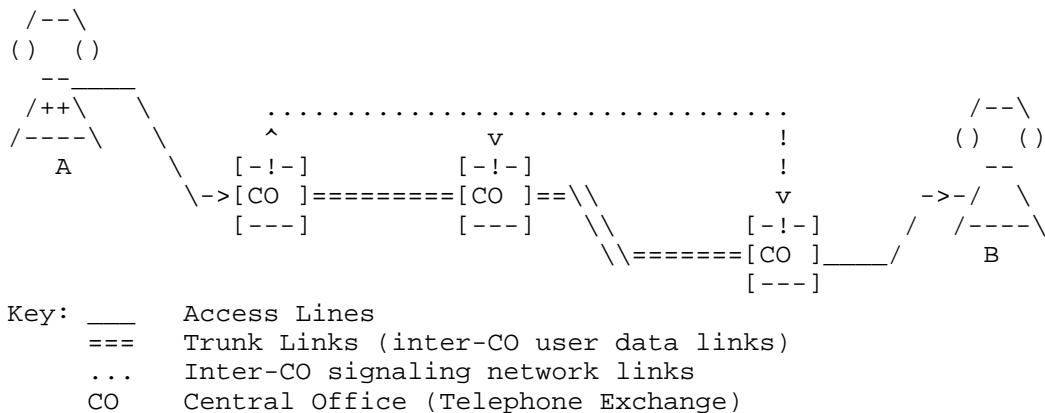
Figure 17

The user presses a sequence of numbers on a telephone handset
(labeled A), and the telephone passes a sequence of digits (either as
pulses or tone pairs) to the Central Office via the access line. The
Central Office contains a processor that will be notified that the
user has made a request and the digit string that is the sole
parameter of the request. This digit string is taken to be the unique

address of an access line connected either to itself or to another
Central Office. There is a hierarchical addressing scheme, so that
the digit string can be parsed easily. A call request to a terminal
(labeled B) connected to a remote Central Office can be routed by
examining the digit string passed; the Central Office will extract
the part of the passed address that corresponds to the remote Central
Office in question, and can route the request onward, forming an
inter-Switch call request and passing it via the signaling network.
At the same time it will allocate one of its available transmission
channels towards the remote Central Office.

11.1.2 Intelligent Network

This scheme has been used since the 1950s, and suffices for the
majority of calls. However, there are a range of other services that
can be (and have been) provided, enhancing this basic call
processing. Freephone or Premium Rate services (1-800 or 1-900
services) are good examples of the supplementary services that have
been introduced. Apart from the important feature that the cost of
these calls is varied so that the caller does not pay for a free-
phone call, or pays an extra charge for a premium rate call, they
have the similarity that the number dialed must be translated to
arrive at the "real" address of the destination terminal. They are
known as number translation services, and make up the bulk of all
supplementary services delivered today.

These were originally programmed into each Central Office, but the
complexity of maintaining the data tables on each processor grew
cumbersome, so a more general solution was sought. After a
considerable gestation period, the eventual solution was the
Intelligent Network. This takes the separation of Central Offices and
the network links interconnecting them a stage further.

The Central Offices are considered to provide the Call Control
Function (CCF). In addition, the Service Switching Function (SSF) is
provided to "enhance" the operation of these Switches by detecting
when a particular request has been made (such as by dialing 1-800).
If this pattern is detected, the equipment implementing the SSF will
send a specialized request message over the signaling network to a
separate computer that implements the Service Control Function (SCF).
This entity is responsible for querying service specific data (held
in a unit providing the Service Data Function, or SDF), performing
any digit translations necessary, and sending the details of how to
proceed back to the SSF, where they are obeyed and the call is put
through to the "real" destination. In many implementations, the SDF
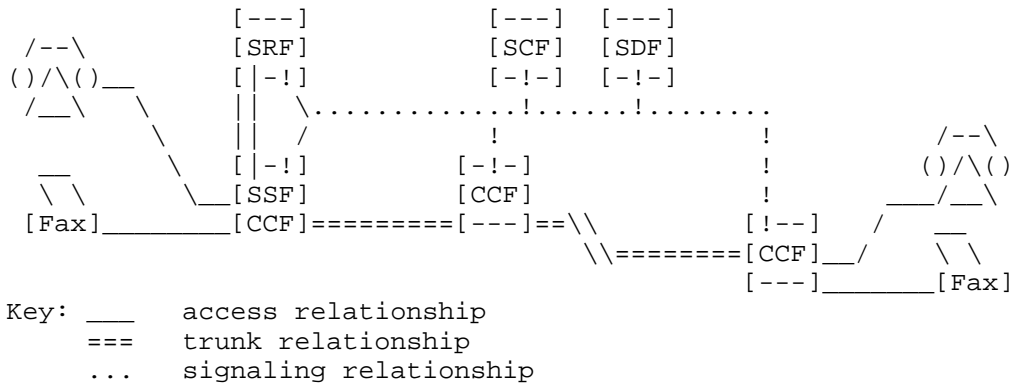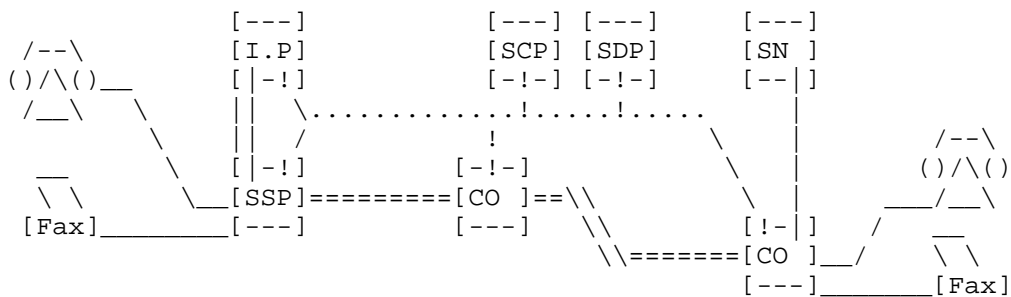is closely coupled to the SCF. This configuration is shown in Figure
18.

```
              [---]           [---]  [---]
   /--\       [SRF]           [SCF]  [SDF]
  ()/\()__    [|-!]           [-!-]  [-!-]
   /__\   \   ||   \..............!.....!.......
        \   \  || /            !              !         /--\
   __     \  [|-!]      [-!-]                 !        ()/\()
   \ \     \__[SSF]      [CCF]                 !         ___/__\
  [Fax]_____[CCF]========[---]==\\         [!--]    /    __
                                   \\=======[CCF]__/      \ \
                                           [---]_____[Fax]
   Key: ___    access relationship
        ===    trunk relationship
        ...    signaling relationship
```

Figure 18

The advantage is that there can be a much smaller number of physical
units dedicated to the SCF, and as they are connected to the
signaling network they can be contacted by, and can send instructions
back to, all of the units providing the SSF and thus the CCF.

In another enhancement, a separate entity called the Special Resource
Function (SRF) was defined. Equipment implementing this function
includes announcement units to play recorded messages (for example,
prompts to enter digits) to callers. It will also include the tone
decoders needed to capture any digits pressed by the caller in
response to the prompts. It is connected to the rest of the PSTN
usually via trunk data links. It will also include a signaling
connection (directly or indirectly) back to the SCF, via the PSTN's
core signaling network.

As an example of the way that these different functional entities
interact, the SCF can ask an SSF handling a call to route the caller
temporarily through to an SRF. In response to instructions sent to it
from the SCF over the signaling network, the SRF can play
announcements and can collect digits that the user presses on their
terminal in response to prompts they are played. Once these digits
have been collected they can be passed on to the SCF via a signaling
message for further processing. In normal operation, the SCF would
then ask the SSF to dissolve the temporary connection between the
user's terminal and the SRF. This allows the collection of account
numbers or passwords (or PINs) and forms the heart of many "Calling
Card" services.

This pattern of user interaction is also used in a wide variety of
other services where extra account information and PINs are needed.
They are collected as just described and can be checked against the
correct values stored in the service database prior to allowing the
call to proceed.

The Intelligent Network functional entities can be realized as
physical units in a number of different combinations. A common
configuration is shown in Figure 19.

```
                      [---]              [---] [---]     [---]
      /--\            [I.P]              [SCP] [SDP]     [SN ]
     ()/\()__         [|-!]              [-!-] [-!-]     [--|]
      /__\   \        ||   \.............!.....!.....     |
            \  \      ||   /             !              \  |        /--\
      __     \     [|-!]          [-!-]                  \ |       ()/\()
       \ \    \__[SSP]========[CO ]==\\                   \ |    ___/__\
     [Fax]_____[---]              [---]  \\            [!-|]   /   __
                                             \\======[CO ]__/     \ \
                                              [---]_____[Fax]
```

    Key: ___   Access Lines
          ===   Trunk Links (inter-CO user data links)
          ...   Inter-CO signaling network links
          SSP   Service Switching Point - a unit that implements the
                  Service Switching Function
          CCP   Call Control Point - a unit that performs call control
                  functions.
                  This is normally a kind of Central Office (shown as CO
                  above)
          SCP   Service Control Point - a unit implementing the Service
                  Control Function. NOTE that this is connected to the SS7
                  Network and uses this connection for all of its
                  communications.
          I.P   Intelligent Peripheral - a unit that contains specialized
                  resources (like announcement units, tone decoders).
                  In effect, it implements Special Resource Functions.
          SN   Service Node

                               Figure 19

This diagram also shows a unit called a Service Node, or SN. This
contains components that realize all of the operational Intelligent
Network functions (SSF, SCF, SDF, and SRF). It is sometimes more
convenient to have all of these elements in one node (for example,
for operations and maintenance reasons), particularly within smaller
PSTNs or where there is a relatively low level of requests for

particular services. Another difference is that, as they are all co-
located, proprietary protocols can be used for internal
communication, rather than the full Intelligent Network Application
Part (INAP) protocol used over the core signaling network between
discrete units. It also differs from the "unbundled" approach in that
it is connected to the COs within a PSTN as a peripheral, having only
an access connection to a Central Office; there is no connection to
the core signaling network. Other than this, it operates in a similar
way, and can provide the same kinds of services. Information on the
specification of the Intelligent Network can be found in the ITU
recommendations [1], while two books ([2] and [3]) describe the
system, its history, operation, and the philosophy behind it.

11.2 Call Center Features

A Call Center is a system that allows a company to be organized with
a group of similar individuals (agents), all of whom can either make
calls to, or take calls from, customers. The system distributes
incoming calls to the agents based on their availability and
automates the placement of outgoing calls, selecting an agent to
handle the call and routing the call to them only once the call
request has been made of the PSTN.

The incoming call distribution feature ("automatic call
distribution", or ACD) is usually coupled with a call queuing scheme.
In this scheme, the callers are connected temporarily with an
announcement unit that normally plays music. The calls are treated in
sequence so that (once the caller is at the front of the queue) the
ACD system selects the next available agent and routes the call
through to them.

Another feature connects a customer making an incoming call to a unit
that asks them for some information on the purpose of their call,
selecting the agent to handle the call based on the particular area
of expertise needed; to do this, the agents are further categorized
by their knowledge (or "Skill Set"). If this skill set categorization
is used then by implication there will be separate queues for each of
the skill sets. This user selection scheme can be used independently
of the others. For example these so-called "voice navigation systems"
can be used to select a particular department extension number, based
on the function required by the customer; as such, they can automate
the job of company telephone receptionist in routing incoming calls.

Where possible, the information gleaned from the customer can be
provided to the selected agent, usually via a separate networked
computer connection. Similarly, if an outgoing call is being made to
one of a list of customers, information on the customer and the
purpose of the call can be provided to the agent selected to handle

the call. Such configurations are generally called "Computer
Telephony Integration" or CTI systems. Strictly, a CTI system can be
arranged to handle routing of incoming calls and automation of
outgoing calls only (also known as computer integrated telephony
features), without the agents having access to a network of
computers. However, the business case for combining the telephony
functions of the call center with provision to the agents of
computers with customer information can be compelling.

This is often further combined with a company's order and service
processing computer system. In this case, a call is treated as part
of a business transaction, with the information to be exchanged
captured as fields of a computer form. While such a computer system
is not, strictly, part of a call center, integrating the company
computer system with the call center is very common. This allows the
details of the call to be stored on a centralized database, allowing
further automated order processing, for example. It also allows the
call to be transferred from one agent to another where needed,
ensuring that the new agent has the information already captured.
This might be useful if someone with a different area of expertise
were to be needed to handle the customer's requirements.

Traditionally, Call Centers have been used to support teams of agents
working at a single site (or a small number of sites, with private
telephony trunks interconnecting them). The site Private Automatic
Branch eXchange (PABX) was integrated with a computer system to
provide these features to people at that site. There can be a
business case for provision of such features to distributed teams of
workers as well. In particular, the possibility of providing support
for people working from home has been seen as important. Some of the
Call Center features have been incorporated into public telephone
exchanges or Central Offices (COs) from many manufacturers as part of
their "Centrex" service offerings.

There are practical limitations in providing such features on COs.
Apart from the procedures needed to configure these features for any
telephone line that is to use them, the basic requirement that every
agent must have a connection to the supporting CO can limit its
usefulness. Another approach is to provide Call Center features via
the Intelligent Network. The features might thus be provided over a
Telephone Operator's entire network, and would mean that the Call
Center could be configured centrally while still allowing agents to
be located anywhere within the telephone network. It also means that
the supported company can pay for the Call Center features "as they
go" rather than having a high "up front" cost.

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