ENGINE server network

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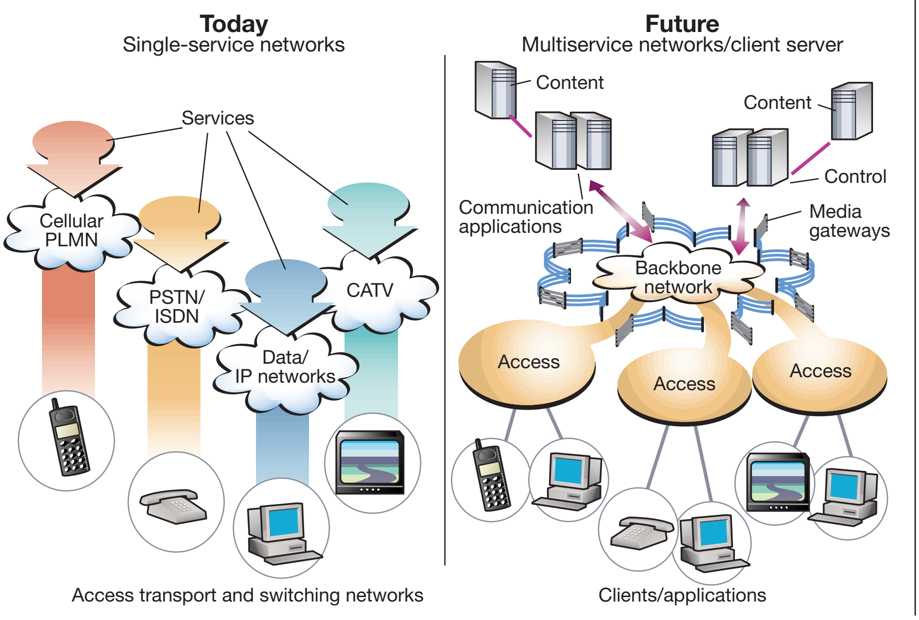
ENGINE—Ericsson's next-generation network solution—allows existing circuit-switched networks to evolve smoothly and rapidly into packet-based multi-service networks. The ENGINE name comprises a complete family of system products as well as various configurations of network solutions that are adapted to meet the needs of different types of operator.

The authors describe the ATM-based ENGINE solutions, putting special emphasis on the server-based solutions. The authors also describe the migration path toward an IP-based solution.

Network trends

Today, the services, switching, and transport of most networks are vertically integrated or intertwined—that is, operators generally have separate telephone, data, and broadcast networks. What is more, few of today's modern networks are integrated or permit unbundling; therefore, considerable efforts are being made to develop standards and equipment that enable integration and unbundling. In the past, it was possible to use common fiber, synchronous digital hierarchy (SDH), and synchronous optical network (SONET) technology as a common transport network for a number of different services. By contrast, the next-generation networks will employ a common switching and transport layer—a connectivity network. This constitutes a move away from vertically integrated networks to horizontal networks, in which transport and switching can be shared by numerous services (Figure 1).

Figure 1
Evolution of the network architecture.



At the edge of the network, the different services are adapted to a common transport technology that is used within the connectivity network. Basically, there are only two transport technologies worth considering: these are the Internet protocol (IP) and asynchronous transfer mode (ATM), which also carries IP for IP services. ATM is a mature technology that gives the right quality of service (QoS) and allows managed provisioned bandwidth—for instance, for local area network (LAN) interconnection. Accordingly, many operators prefer ATM as the common transport technology format for the connectivity network.

To transport IP on ATM, an edge router (ER) is deployed at the edge of an ATM-based connectivity network (Figure 2). Paths through this network can be set up manually by means of permanent virtual connections (PVC) or soft PVCs (SPVC). They can also be set up automatically by means of multiprotocol label switching (MPLS), in which case the edge router functions as a label edge router (LER) and the core ATM switches serve as label switch routers (LSR).

Frame relay technology has been adapted to ATM via a frame-relay interworking function.

For telephony service, the International Telecommunication Union (ITU) and Internet Engineering Task Force (IETF) have drafted standards for splitting the service adapter into a controlling entity (media gateway controller, MGC) and a bearer plane adapter (media gateway, MGW), which adapts to the technology employed in the connectivity network. In the ENGINE architecture, the MGC is called the telephony server (TeS).

Operator benefits

There are several reasons why operators want to migrate to an ATM network and server solution (MGC-MG architecture) for telephony traffic. ATM networks give operators a common switching and transport technology that supports numerous different services. In essence, with ATM, operators gain a multiservice network—on the same network, they can have LAN interconnect via ATM or frame relay, IP-based virtual private network (IP-VPN) using MPLS, telephony traffic, and so on. Furthermore, with packet technology, ATM offers improved use of bandwidth by means of statistical multiplexing.

Another advantage of ATM is that it is a more compact technology than circuit switching, thereby reducing floor-space requirements and power consumption. With a multiservice network built on ATM, operators can both build for the future and reduce present-day operating costs. Finally, it should be emphasized that ATM-based solutions can deliver the same service quality as present-day circuit-switched networks.

With server-based solutions, large service control nodes can be created without constraints from the underlying transport topology. This implies simpler and faster introduction and management of new services—factors that become increasingly important as competition in the telecom market stiffens. Furthermore, the service control layer becomes independent of the technology used in the connectivity network, which minimizes the impacts of introduc-

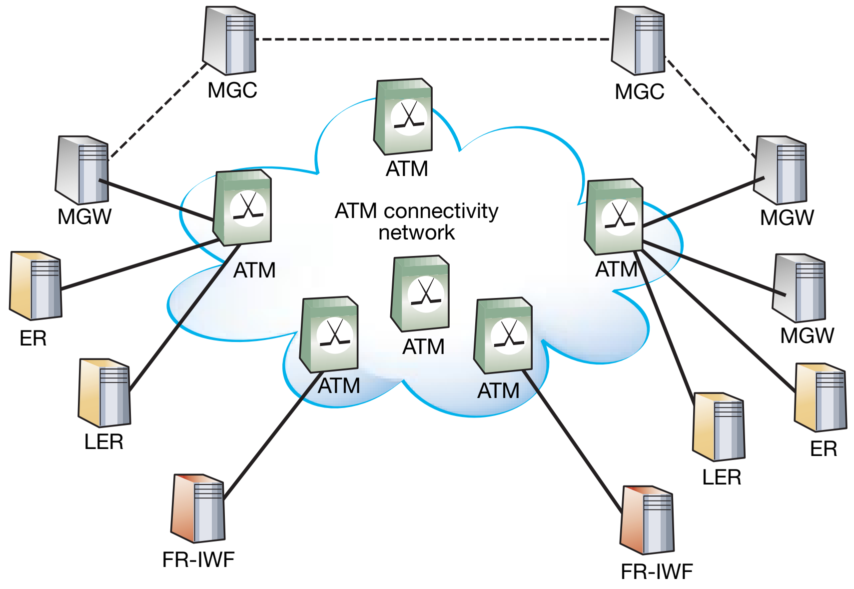


Figure 2
Connectivity network based on ATM—including edge adapters.

BOX A, ABBREVIATIONS

AAL1	ATM adaptation layer 1
AAL2	ATM adaptation layer 2
AINI	ATM internetwork interface
AM	Application module
AMR	Adaptive multirate (code)
AN	Access node
ATM	Asynchronous transfer mode
BICC	Bearer-independent call control
CATV	Cable television
CE	Circuit emulation
CR-LDP	Contraint-based routing - label distribution protocol
DCME	Digital circuit-multiplexing equipment
DEV	Device
DSL	Digital subscriber line
DSS1	Digital signaling system no. 1, used for ISDN access
DT	Dynamic trunking
E1	ETSI 2 Mbit/s interface
ER	Edge router
FR-IWF	Frame relay interworking function
G.711	ITU-T recommendation for PCM coding
GMSC	Gateway mobile switching center
GPRS	General packet radio service
GS	Group switch
GSN	GPRS support node
H.248	ITU-T recommendation for media gateway control
H.323	ITU-T recommendation for packet-based (mainly focusing on IP) multimedia communications systems
HCI	Half-code interface
IETF	Internet Engineering Task Force
IN	Intelligent network
IP	Internet protocol
ISDN	Integrated services digital network
ISUP	ISDN signaling user part
ITU	International Telecommunication Union
ITU-T	ITU Telecommunications Sector
IWU	Interworking unit
LAN	Local area network
LE	Local exchange
LER	Label edge router
LSP	Label switched path
LSR	Label switch router
MBHCA	Mega busy-hour call attempt
MGC	Media gateway controller
MGW	Media gateway
MPLS	Multiprotocol label switching
MTP	Message transfer part
MTP3	MTP, Layer 3
MTP3B	MTP3 on ATM
OA&M	Operation, administration and maintenance
PBX	Private branch exchange
PDH	Plesiochronous digital hierarchy
PLMN	Public land mobile network
PNNI	Private network-network interface
PPP	Point-to-point protocol
PRA	Primary rate access
PSTN	Public switched telephone network
PVC	Permanent virtual connection
Q.1901	ITU-T recommendation for the BICC protocol
Q.2630.1	Signaling protocol for AAL2 connections, also called Q.AAL2
Q.765.5	ITU-T recommendation specifying part of the BICC standard
Q.931	ITU-T recommendation for layer 3 DSS1 signaling
Q.AAL2	See Q.2630.1
QoS	Quality of service
RMP	Resource module platform
RSVP	Resource reservation protocol
SCTP	Stream control transmission protocol
SDH	Synchronous digital hierarchy
SDP	Session description protocol
SE	Switch emulator
SIP	Session initiation protocol
SONET	Synchronous optical network
SPVC	Soft PVC
STM-1	Synchronous transport module, 155 Mbit/s link
SVC	Switched virtual connection
TE	Transit exchange
TeS	Telephony server
TUP	Telephony user part
UMTS	Universal mobile telecommunications system
UNI	User network interface
UTRAN	UMTS radio access network
V5.2	ITU-T recommendation for fixed access network interface
VoDSL	Voice over DSL
VPN	Virtual private network

ing new or enhanced bearer technologies. It also enables operators to use best-of-breed components.

BOX B, MPLS

Multiprotocol label switching (MPLS) is an optimized virtual circuit-oriented switching layer for IP networks. When entering the MPLS network, IP traffic is tagged with an MPLS label in a label edge router (LER). Once inside the MPLS network, the label switch router (LSR) forwards traffic along the label-switched path according to the value of the tag. When the traffic reaches an LER at the edge of the MPLS network, the tag is removed and the packet is forwarded using regular IP techniques. MPLS labeling has been defined for

- ATM networks (where the ATM virtual path/virtual circuit field carries the MPLS label);
- point-to-point IP networks running the point-to-point protocol (PPP); and
- frame relay.

MPLS is a unique marriage of IP and ATM capability that allows ATM networks to become IP-aware and handle Internet traffic with the same scalability as an IP network. MPLS also allows IP networks to offer virtual circuit-like structures.

Another advantage of server-based solutions is that media gateways, which can be located as remote switches, can be controlled by a single media gateway controller. This simplifies operation, administration and maintenance (OA&M), since the complex logic and call-related data can be centralized. It also yields more optimized routing of the actual bearer path for call forwarding and other service actions.

Available ENGINE products

Several products from Ericsson's ENGINE program are already being used commercially or in trials. There are currently three ENGINE solutions:

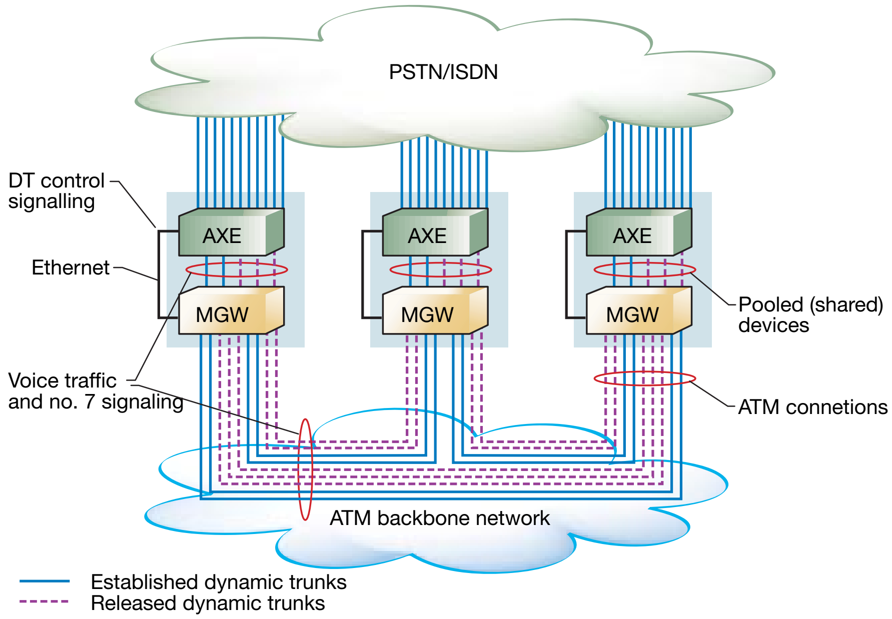
- dynamic trunking solution (ENGINE trunked network);
- the telephony server single-domain solution (ENGINE Bridgehead); and
- the hybrid switch solution (ENGINE switched network).

Dynamic trunking solution—ENGINE trunked network

Dynamic trunking enables operators to set up and release resources dynamically according to traffic load. The function is implemented in the AXE narrowband switch and the AXD 301 ATM switch (the latter serves as a media gateway, MGW, or edge adapter), which operate together as one node. Nodes are interconnected over an ATM backbone network via the media gateway (Figure 3).

To reduce hardware and maintenance costs in the node, the E1 and STM-1 interfaces between AXE and AXD 301 are pooled—that is, they are shared for all destinations. This means that routes need not be dimensioned individually. A dynamic trunk in the ATM network consists of several on-demand switched virtual connections (SVC), which are automatically established and released according to traffic load in AXE.

Figure 3
The ENGINE trunked network.



Telephony server single-domain solution—ENGINE Bridgehead

The first version of the telephony server solution is a single-domain solution—that is, it does not use bearer independent call control (BICC) signaling to interconnect different domains. The telephony server solution separates call control and bearer control in the network. The two main parts of the solution are the telephony server and the media gateway (Figure 4).

The telephony server is divided into two parts:

- AXE provides the telephony functionality. This part contains the software for call control, which includes number and routing analysis. It also contains all telephony functions used in the circuit-switched telephone networks, such as charging and accounting services, and different types of signaling support. The use of AXE as a component in the telephony server guarantees smooth migration to an ATM-based multiservice network with full transparency of existing telephony functions.
- the switch emulator (based on the AXD 301 platform) provides the interface between the call control part and the MGW, and controls the switching resources that are implemented in the MGW. Ericsson's own media gateway control protocol is used, since H.248 standardization had not been finalized at the time of implementation.

The media gateway (which is also based on the AXD 301 ATM switch) performs switching and media interworking functions between the circuit-switched and

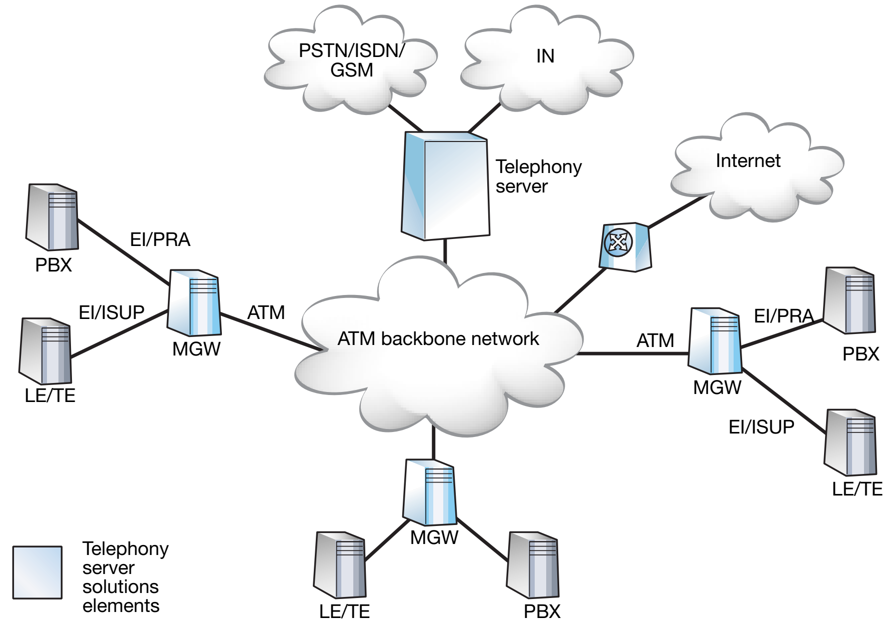


Figure 4
ENGINE Bridgehead.

ATM domains. It also establishes the ATM voice bearer connections (switched virtual connections) via standardized ATM signaling interfaces. Using circuit emulation, the media gateway is able to connect local exchanges, transit exchanges, and private branch exchanges (PBX).

Hybrid switch solution—ENGINE switched network

The hybrid switch solution, which is a node solution, provides a high-capacity core node that supports telephony and data. The node is built up from a combination of AXE and AXD 301. The ATM switching resources for telephony are controlled via a control link with an internal system protocol. The circuit-switched connections are connected to circuit-emulation boards; on the ATM side, preconfigured virtual connections are used. The ATM virtual channels solely take up capacity in the ATM network when connections have been established for telephony traffic. ISDN signaling user part (ISUP) signaling between the nodes is used for establishing connections and controlling calls. For datacom services, all existing AXD features are supported and can operate simultaneously with telephony service.

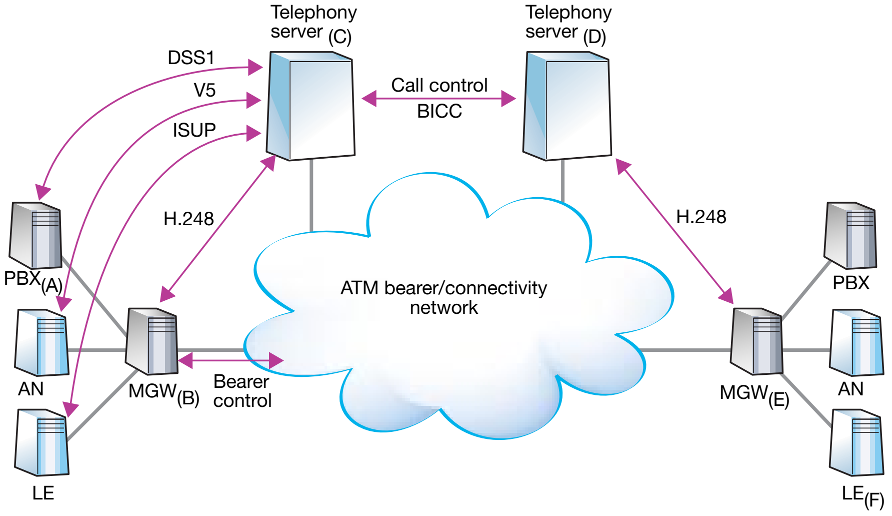
ENGINE Integral

Like ENGINE Bridgehead, ENGINE Integral gives all the advantages mentioned thus far, including cost reductions thanks to the consolidation of networks, reduced power consumption, and flexible network architectures. However, it also satisfies operators' requirements for immediate and future networks. ENGINE Integral offers communication between domains using the ITU-T standardized BICC interface, and controls media gateways by means of the standard H.248 protocol. ENGINE Integral also introduces echo cancellers in the media gateway.

Figure 5 shows the basic network configuration for telephony. The telephony server handles call logic, and the media gateway manages media adaptation, using ATM adaptation layer 1 (AAL1). Local exchanges (LE), access nodes (AN) and PBXs are connected to the media gateways. Transit and tandem exchanges can also be connected to them. Signaling associations are set up between the telephony server and local exchange, access node, and private branch exchange. To understand the architecture and call and connection set-up, let us follow a call from a private branch exchange (A) to a local exchange (F).

The primary rate access (PRA) interface is terminated in the media gateway (B), but the signaling channel on TS 16 is sent transparently via the ATM network to the TeS on a permanent virtual connection (PVC) using AAL1 circuit emulation (CE). The

Figure 5
ENGINE Integral.



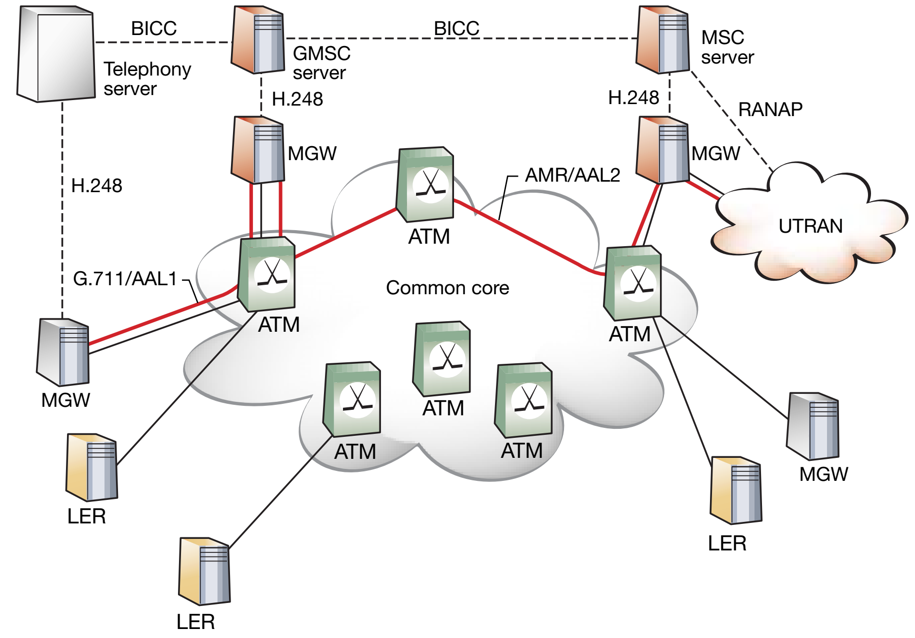


Figure 6
Common bearer network for fixed and mobile telephony.

main advantages of using circuit emulation on AAL1 are that existing signaling terminals can be used and that the media gateway is transparent to the different types of signaling from the circuit-switched network (provided that outband signaling is used).

In our example call, the private branch exchange sends a call set-up message to the telephony server (C). The telephony server (C) busy-marks the timeslot used on the PRA interface and analyzes the number of the called party. Since this particular call will terminate in the locat exchange (F) of another server domain, a call set-up message is sent using BICC signaling to the telephony server (D). When the telephony server (D) receives the call set-up message, it asks the media gateway (E) to provide a bearer connection identifier and ATM address.

The reference and address are returned in the BICC signaling to the telephony server (C), which orders the media gateway (B) to set up a bearer to the media gateway (E) using standard ATM signaling procedures. After the bearer has been set up, the media gateway (E) uses the bearer connection identifier as a way of binding the bearer to the call and notifies the telephony server (D) that the bearer has been established.

The telephony server (D) then orders the media gateway (E) to through-connect and send a call set-up message using ISUP or telephone user part (TUP) to the local exchange (F). As with PBX signaling, the ISUP signaling links in the message transfer part (MTP) are transported transparently in the ATM network using circuit emulation on AAL1. The remainder of the call set-up procedure is the same as for a regular narrowband call.

The scenario we have just described uses the forward-connection set-up method, but ENGINE also supports the backward-connection set-up method.

Since ENGINE introduces a multiservice network, it can also be used as a bearer network for mobile networks. For example, when deploying general packet radio service (GPRS), operators can use the bearer network to interconnect cellular access networks with serving GPRS support nodes (GSN), and to interconnect serving GSNs with gateway GSNs. In a large take-up of GPRS, the bearer network will primarily be used between GSNs, since the serving GSN will be physically close to the interconnection point of the mobile access network.

When the universal mobile telecommunications system (UMTS) is introduced, the bearer network can be used to interconnect mobile telephony MGWs. In this way, there is convergence on the bearer layer between fixed and mobile networks. At the server

BOX C, BICC

Bearer-independent call control (BICC) is a standardization item within SG 11 in ITU-T. BICC will be used as a call-control protocol in telephone networks when they evolve to support different transport technologies with the same (ISUP-based) service set. BICC can be used to separate call and bearer signaling. BICC (which is based on ISUP) and ISUP are very similar. In BICC, the functions for controlling the bearer have been removed and ways of binding functions between BICC and bearers have been added. Several methods can be employed to transport BICC signaling, including MTP3, MTP3B and IP.

BICC CS1 emphasizes ATM as the bearer technology. Both AAL1 (with UNI, PNNI, DSS2, and B-ISUP signaling) and AAL2 (with Q.AAL2—also called Q.2630.1) signaling have been considered. BICC CS1 was determined in December 1999. For BICC CS2, relations to IP as the bearer will be specified as well as the use of H.248 as a protocol between the call-control and bearer-control entities. The determination of BICC CS2 is planned for year-end 2000. BICC CS1 is specified in Q.1901, and Q.765.5

level, the two remain separate, which means there will be few logical and physical points where fixed and mobile telephony interconnect (Figure 6).

Telephony server

The telephony server in the ENGINE Integral solution is similar to that used in the ENGINE Bridgehead solution, but with the following enhancements:

- communication between servers is possible over the ITU-T standardized BICC interface;
- the media gateway control has been adapted to H.248;
- the call-handling capacity has been increased; and
- multinational telephony server functions have been added.

The telephony server consists of AXE and AXD 301 systems. The AXE part handles call control and telephony functions, whereas the AXD 301 part provides the interface to the ATM network and contains a switch emulator (SE) and media gateway function that connect to an integrated resource module in the server (Figure 7).

The switch emulator is controlled by AXE from the resource module platform (RMP), which was introduced in AXE together with the application-oriented architecture in order to separate control of the switch from applications. The resource module platform can thus be extended to control a separate ATM bearer network. An internal system protocol, used between AXE and AXD 301, controls the switch emulator.

The switch emulator executes the H.248 protocol and distributes the control links to the media gateways. The H.248 links are transported over the ATM network. A resource module has been integrated into the server to handle announcements, conference bridges, signaling terminals, and so on. The group switch in AXE, which is used for connecting these devices (DEV), enables the reuse of existing devices in AXE. This translates to lower investment costs when operators upgrade AXE to serve as a telephony server.

For communication between the server domains, the BICC protocol is used for call-control signaling, and ATM signaling is used for bearer-control signaling. The open-standard interface enables operators to build multivendor networks.

The AXE system contains the software for call control (number and routing analysis). It also contains the telephony functions used in circuit-switched telephone networks (charging and accounting services, and different kinds of signaling support). The use of AXE as a component of the telephony server guarantees smooth migration to an ATM-based multiservice network with full transparency of existing telephony func-

BOX D, H.248

The H.248 protocol—the result of cooperation between organizations in the telecommunications and data communication sectors—led to a joint effort between the ITU-T SG-16 and the IETF. The protocol is used to control

- switching;
- media-stream manipulation;
- transport technology adaptation; and
- other functions residing in media gateways.

H.248 can be employed in various cases that have media gateways in their architecture. To cope with such variety, use-case packages have been defined within H.248—for example, a package for connecting circuit-switched ISUP trunks where the attributes of the ISUP trunk terminations can be described.

The protocol, drafted by the ITU, is in the last-call stages at the IETF. The final draft was approved in June at a meeting of the ITU (SG 16). Other industry organizations, such as 3GPP, Tiphon, the Multiservice Switching Forum (MSF), and the VoDSL/xDSL Forum, have also adopted this protocol.

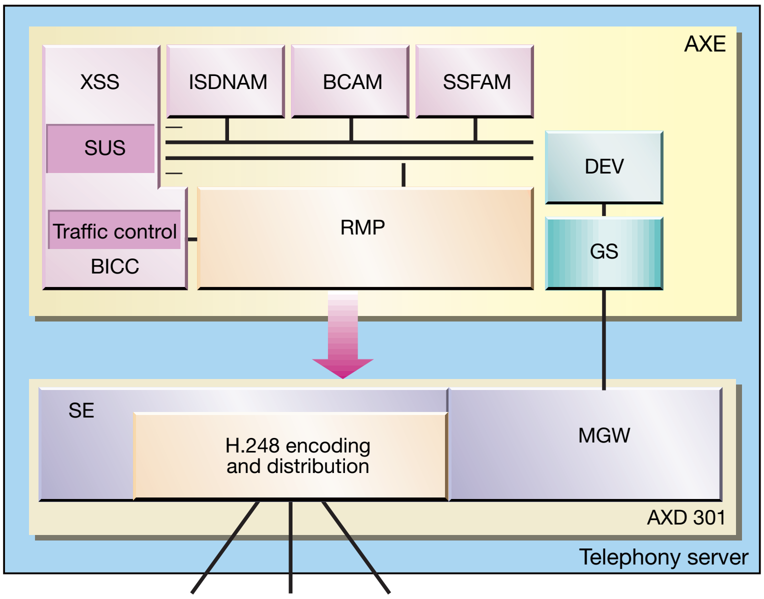


Figure 7
Telephony server architecture.

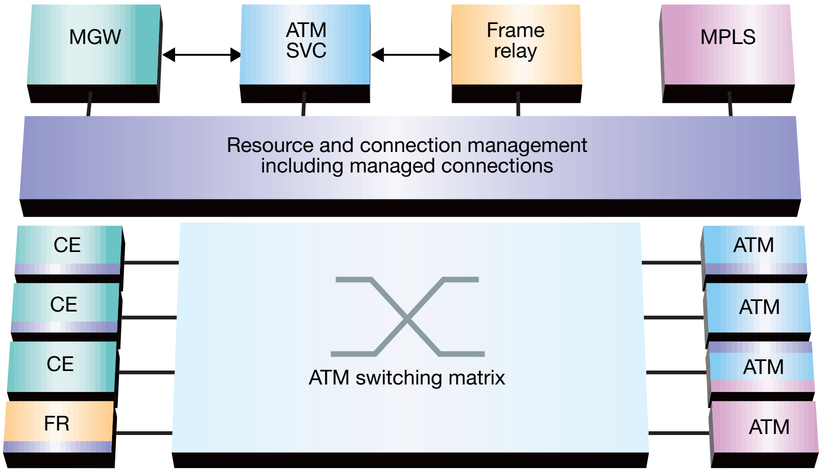


Figure 8
The AXD 301 multiservice architecture.

tions. Another advantage of using the AXE system as a basic component in the server is that it can be combined with the handling of normal circuit-switched traffic. This capability is used for connecting trunks with inband signaling directly to the group switch. Thanks to the widespread deployment of AXE, a multitude of market variants of signaling systems is supported, which also means that the complexity in the media gateways can be reduced.

The call-handling capacity of the telephony server in the first release of ENGINE Integral is estimated to be around 2 mega busy-hour call attemps (MBHCA). This capacity will increase in subsequent releases.

With duplicated processor systems in both AXE and the AXD 301, the server offers carrier-class availability.

The multinational telephony-server feature allows a telephony server to control media gateways located in different countries, which is a very cost-effective solution for global operators.

Media gateway

The media gateway is built as an application on top of the AXD 301 ATM switching platform. The architecture of the AXD 301 system allows simultaneous support of several different service applications (Figure 8), each of which controls its own partition of interface resources and can be guaranteed a configurable share of processing resources.

All communication between the telephony server and the media gateway is handled by the H.248 protocol. The heart of the media gateway application is the H.248 context and termination manager (Figure 9), which coordinates communication between the telephony server and other functional entities of the media gateway, such as access control functions, bearer control functions, and switch control functions.

Access control functions

Via access control functions, various kinds of access can be put under the control of the telephony server. The access type is more or less transparent to the media gateway, which provides interfaces to access nodes, network trunks, and ISDN primary rate accesses. The physical access might consist of a single E1 plesiochronous digital hierarchy (PDH) interface or multiple E1s multiplexed in an SDH STM-1 interface. The individual timeslot of the E1 interfaces is usually mapped to ATM using AAL1 64 kbit/s circuit emulation, but $n \cdot 64$ kbit/s AAL1 circuit emulation is also supported—for instance, for leased lines in the ATM network.

Bearer control functions

Bearer control functions are used to establish the 64 kbit/s ATM bearer connection between two media gateways. The bearer connection is established from one media gateway to another by means of regular

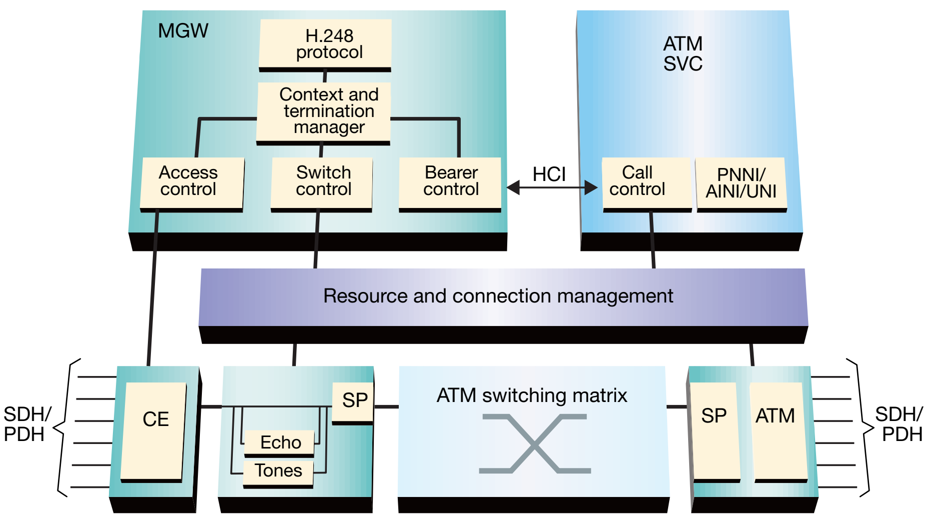


Figure 9
The AXD 301 media gateway architecture.

ATM signaling procedures using the ATM address that was provided by the server via the H.248 protocol. The ATM signaling also carries the connection identifier provided by the server via the H.248 protocol, as the means of correlating the ATM bearer connection with a particular call. Since the bearer control function uses the internal half-call interface in AXD 301, the bearer connections to be established can use any of the signaling protocols supported by AXD 301, such as the user network interface (UNI 4.0) and the private network-to-network interface (PNNI).

Switch control functions

Switch control functions in the media gateway permit switch paths to be connected and disconnected between different access ports and ATM bearer ports. It also supports the connection of auxiliary resources, such as tone senders, echo cancellers, and so on. The switch control function enables the media gateway to function either as an ordinary originating or terminating media gateway and as a relay media gateway with ATM bearers on both sides—when the telephony server controls the routing of the ATM bearer connection.

The media gateway application also implements the concept of logical media gateways, each of which is controlled by an individual H.248 association and solely serves on a subpartition of the resources allocated to the media gateway application. This allows one AXD 301 system to function as several logical media gateways, where each logical media gateway is controlled by a different telephony server.

As mentioned above, the AXD 301 system can simultaneously support several service applications. When serving as a media gateway for voice services, it can also serve as an edge or core ATM switch, frame-relay node, or MPLS router. Thus, one AXD 301 can connect interfaces for voice, ATM, frame relay, and IP services that share the same ATM backbone network for connectivity.

The media gateway application in AXD 301 scales from a few thousand to over a million 64 kbit/s channels with tone sending and simultaneous echo-cancelling capabilities for every channel.

Further enhancements of ENGINE Integral

The ENGINE Integral solution for telephony is an ATM-based AAL1 solution. Subsequent development will address AAL2, in order to handle the transport of compressed voice as a way of

- replacing digital circuit-multiplexing equipment (DCME);
- interconnecting with UMTS on AAL2; and
- interconnecting with voice-over-DSL (VoDSL) access.

Although ENGINE Integral already supports IP transport, further enhancements will be made. In terms of telephony, mi-

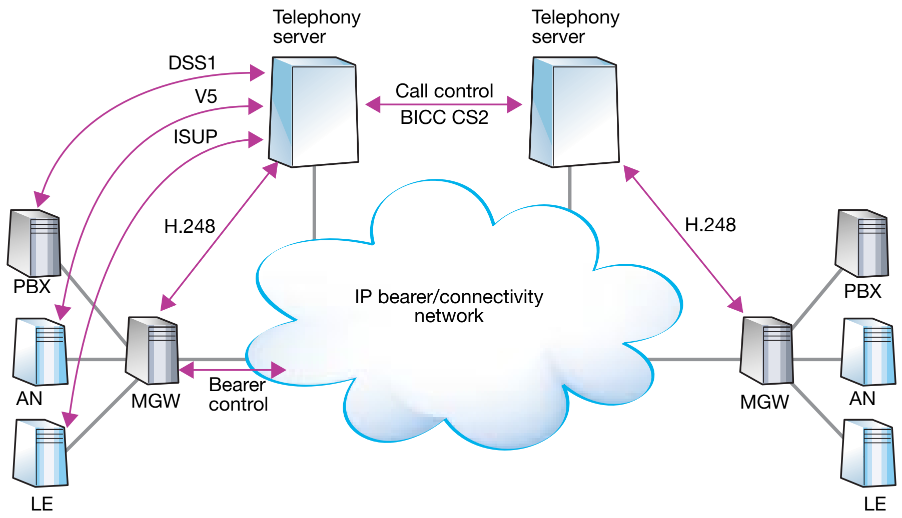


Figure 10
The ENGINE IP solution.

gration from media and signaling transport on ATM to media and transport on IP will be considered.

Migration to an IP-based bearer network for telephony

When considering a packet-based network for telephony service, operators have a choice between ATM or IP bearer networks. ATM networks—originally having been specified with voice service in mind—currently offer more mature technology. However, over the long term, in light of future data and multimedia services, IP networks could become a natural choice. Obviously, operator investments in ATM networks should continue to play a role when telephony service is migrated from ATM to IP—when volumes and revenues from data and multimedia services increase.

Today's IP networks are mainly used for data-oriented services, but new architectures are being standardized to support voice and multimedia services that include a voice component. The voice service in these new architectures is not the same as present-day telephony service, since it does not support all the existing functions of current telephone networks. The new architectures for voice and multimedia networks are mainly based on the H.323 and session initiation protocol (SIP) standards.

The ENGINE migration to an IP-bearer network can be based on the same architecture as the ATM-based solution, in order to preserve existing telephony services. This is different from the H.323 and SIP architectures, which are multimedia-oriented. The expressions IP telephony and telephony-over-ATM/IP are used to distinguish the voice service in these architectures. Telephony-over-ATM/IP indicates a solution in which all existing telephony functionality in the circuit-switched networks is preserved in the new ATM or IP bearer network. IP telephony allows basic calls over an IP network, including some of the services that exist in circuit-switched telephone networks. IP telephony can be a service of its own or the component of a multimedia service.

Ericsson is confident that telephony-over-ATM/IP and the H.323 and SIP architectures will evolve separately. Therefore, the ENGINE telephony-over-IP solution is being based on the architecture defined in BICC standardization. Notwithstanding, the SIP and H.323 architectures and the telephony-over-ATM/IP type of architecture must be able to interwork with one another.

As is done in the telephony-over-ATM solution, the telephony-over-IP solution will use media gateways to connect access nodes, PBXs, local exchanges, and transit exchanges including interconnects to net-

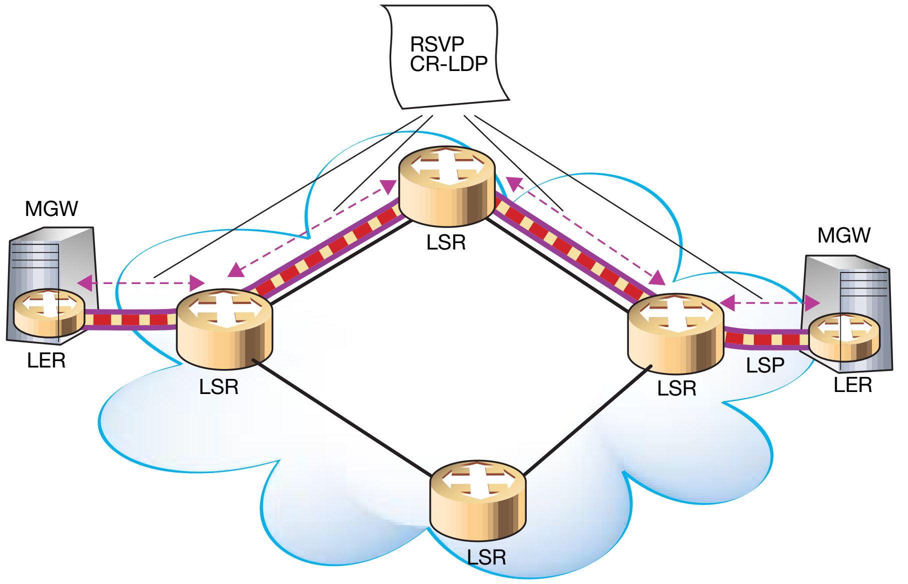


Figure 11
Bearer path between two media gateways over MPLS.

works from other operators with circuit-switched networks.

The transport of the incoming narrowband protocols and of the BICC signaling and the H.248 protocol is different from that of the ATM solution. For the transfer of signaling information, a new standard for signaling transport in IP networks is being developed by the IETF. This standard, called SIGTRAN, uses the stream control transmission protocol (SCTP). Control protocols, such as ISUP, Q.931, and V5.2, are transferred on top of SCTP. Bearer control and set-up differ substantially from the ATM-based solution. In the standardization, it is proposed that a new protocol, based on the session description protocol (SDP), be tunneled in H.248 and BICC between media gateways.

Different ways of guaranteeing the quality of telephony service are being studied by standardization organizations. A very promising approach—employed by the ENGINE IP solution—uses a combination of MPLS and differentiated services. Telephony traffic is transported through the IP network by means of MPLS paths, which are used as voice "trunks" between media gateways (Figure 10). Each trunk (LSP) carries several voice connections. The LSPs can be preconfigured, set up dynamically or dynamically modified to provide bandwidth according to traffic load on the LSPs.

Conclusion

There is a growing trend toward the transport of voice on packet networks. ATM technology is well suited for voice traffic, since it provides on-demand bandwidth connections on a large scale. In combination with the guaranteed quality of service that ATM brings to real-time services, this makes ATM an ideal choice for transporting packet voice.

A multiservice network built on ATM switching for voice, video, and data services gives operators several cost advantages.

- The ENGINE program ensures smooth migration into future packet-based multiservice networks. Products within the program are available today, and in service.
- ENGINE Integral provides a full-fledged server solution based on standards such as BICC and H.248. This allows operators to chose equipment from different vendors. Since the telephony server is based on AXE, the telephony server solution encompasses a wide range of market-specific telephony functions.
- Future enhancements of ENGINE Integral will provide telephony server solutions with IP transport. This will include interworking and the ability to upgrade from the ATM-based ENGINE Integral solution.