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Traffic Trading in the Competitive International Voice Market

Stefano Bregni and Maurizio Dècina, Politecnico di Milano
Giacomo Bruzzi, ICT Consulting S.p.A

ABSTRACT

This article highlights the trends that are reshaping the international long distance voice market. Our analysis focuses on voice exchanges, namely, electronic marketplaces where huge volumes of ILD voice traffic are traded and routed. Various existing models of voice exchanges are compared: Arbinet, which started in 1996, providing TDM circuit-switched interconnection; and newer exchanges, based natively on VoIP, namely, Voice Peering Fabric by Stealth, Infiniroute, XConnect, and Arena by Interoute. The evolution of interconnection mechanisms and the consequent impact on the role of exchanges are discussed. The Calling Party's Network Pays and the Bill & Keep interconnection commercial models are presented, and we discuss issues for their applicability. Per-volume and flat-fee charging models for interconnection between operators are compared. Finally, the impact of these models on future business scenarios of voice exchanges is analyzed. The way in which call termination models will change in the future is uncertain. Regulatory aspects, service-specific requirements, country-specific issues, and market dynamics must all be taken into consideration prior to establishing the interconnection regime that will be widely adopted in the future.

INTRODUCTION

During the last 15 years, the annual growth in volume of international long distance (ILD) voice traffic has been just below 15 percent (although, 25 percent in 2000) [1]. At the same time, the average price of international switched calls declined every year sinking to –24 percent from 1996 to 2002. Thus, as shown in Fig. 1, the overall ILD business was affected by severely declining revenues. Since 2004, the price decline slowed down to about –7 percent, with traffic growth offsetting price reduction and making the ILD overall turnover grow. However, volume growth is slowing down, gradually reducing the overall turnover growth.

Many customers have become familiar with and use Voice over Internet Protocol (VoIP) to place long distance calls at low or zero cost. VoIP traffic has started increasing as fast as the traditional circuit-switched telephone traffic, based on time-division multiplexing (TDM). Figure 2 shows TDM and VoIP (excluding computer-to-computer calling) ILD volume growth from 1999 to 2006. In 2008, VoIP traffic accounted for 37 percent of the total ILD traffic, compared to a 27 percent share in 2007. In addition to this trend, Skype (http://www.skype.com) traffic accounted for 5 percent of the total international traffic in 2006.

Yet, the destination of a large portion of VoIP-generated calls is still in traditional TDM networks. Both in the United States and Western Europe, operators are replacing TDM very slowly or not at all. Their customers are able to place calls using VoIP over broadband access, but they still own (and pay for) a traditional, fixed telephone line, where they typically receive calls, however generated. In addition, most VoIP-generated traffic is bound to developing countries, where calls are terminated on circuit-switched networks.[1]

Prior to the deregulation of global telecommunications, state-owned, monopoly operators exchanged off-net calls and settled traffic imbalances with payments based on bilateral agreements, called *bilaterals*. Traditionally, the buying and selling of international wholesale services was done at major industry events or operator-to-operator. Buyers purchased termination for calls to an entire country at one price (the so-called *country-proper* price).

After deregulation, the number of carriers increased exponentially, and the bilaterals model was challenged by the concept of *competitive termination*, that is, the possibility for an operator in a country to buy call termination from another (intermediate or destination) operator in another country in the competitive market, which typically provides lower unit-price and more price-quality options. As the number of players boomed, new carriers competing in a geographical area differentiated their prices for terminating calls below the country-proper price. The dramatic increase of mobile traffic further boosted the need for competitive termination. The mix of bilateral and wholesale termination in 2007 was around 50–50 [1].

This article highlights trends reshaping the business of the ILD voice market. Our analysis is centered on voice exchanges, namely, electronic

Some of the content of this article was previously presented in the paper "Minutes Trading in the International Long-Distance Voice Market," by S. Bregni, G. Bruzzi, and M. Dècina, which appeared in Proc. IEEE GLOBECOM 2006*, San Francisco, CA, Nov. 2006.*

[1] *The term* call termination *indicates the routing of telephone calls from one operator to another. The* called party *is the termination point. The* calling party *is the originating point.*

marketplaces where huge volumes of ILD voice traffic are traded and routed between operators. Various existing models of voice exchanges are compared, for both TDM and VoIP traffic. The evolution of interconnection mechanisms is discussed. Commercial and charging models of interconnection are compared, assessing their impact on future business scenarios for voice exchanges.

Voice Exchange Models

During the nineties, the concept of *voice exchange* was conceived, based on replacing the traditional one-to-one infrequent trading of ILD voice traffic with fast day-by-day trading. The voice exchange, modeled on the financial exchange concept, allows buyers and sellers to place their bids online, closing transactions when prices match. Optionally, an exchange manages settlements between buyers and sellers, underwriting credit risk, and relieving the sellers of bad-debt risks. In addition, the voice exchange can even route calls to a destination, acting as a transit network.

Trading mechanisms take into account the "quality" of routes, in terms of *routable minutes per call attempt* on the route. To make this effective, routing features must be integrated in the exchange, exempting buyers and sellers from the burden of managing the complexity of calling party codes. Integrated routing enables assessing the true economic value of a traded route by considering its *answer-seizure ratio* (ASR), that is, the percentage of incoming calls that are actually answered through the traded route.

A Paradigm Example: The Arbinet Voice Exchange

Arbinet Inc. (http://www.arbinet.com), quoted on NASDAQ (ARBX), developed a voice exchange in 1996. Arbinet acts as a hub for international competitive voice traffic: trading, routing, and settling, with the goal of reducing the complexity of the many-to-many web of inflexible bilateral agreements with an adaptable one-to-many relationship. Arbinet claims to have (mid-2008) more than 1000 voice members including approximately 75 percent of the world's 40 largest international carriers and eight of the world's prepaid service providers, with $14.4 \cdot 10^9$ minutes traded, routed, and settled in 2007 (+14 percent).[2]

Arbinet makes money from the voice-minute exchange business by charging a transaction fee. Income from buyers is balanced by out-payment for traffic termination to sellers. Net income is thus made of fee revenues.

Originally, Arbinet offered anonymous trading only. More recently, many members started requiring non-anonymous termination for segments of their voice traffic. Launched in 2005, a new service was added by Arbinet, giving members a direct connection to fixed and mobile networks, allowing buyers to purchase routes from known network operators.

Members are connected to a worldwide backbone in one or more points, through standard E1 ports and signaling protocols. Backbone nodes currently are in Los Angeles (two), Miami, New York, London (two), Frankfurt, and Hong Kong.

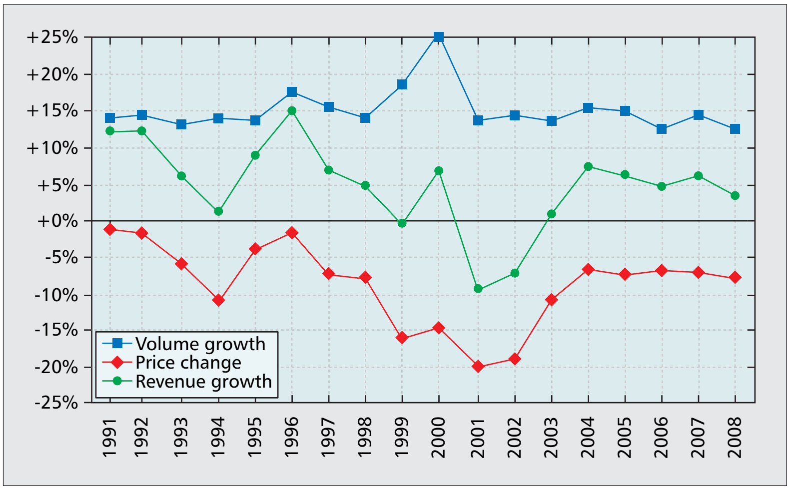

Figure 1. *Rates of annual volume growth, price chang,e and revenue growth in the ILD voice market [1, 2].*

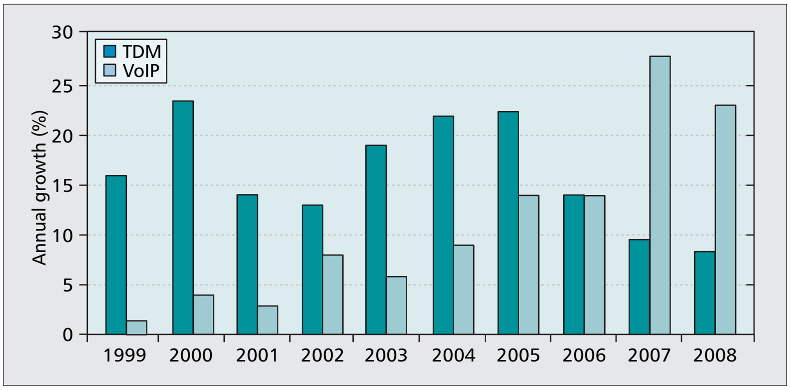

Figure 2. *Rates of VoIP vs. TDM annual volume growth in the ILD voice market (excluding computer-to-computer calling) [2].*

Newer VoIP Exchanges

Because VoIP created the end-user expectation of low- or zero-cost voice calls, pure VoIP service providers expect to avoid traditional interconnection schemes for voice termination. Then, a new breed of VoIP exchanges emerged in the last few years, allowing VoIP providers to bypass bilateral agreements and trading mechanisms in terms of voice minutes.

The major player is Voice Peering Fabric (VPF), a service of Stealth Communications (http://www.thevpf.com) launched in late 2003, operating only in the United States and with one point of presence in London. VPF is a distributed layer-2 Ethernet network with the purpose of exchanging VoIP traffic, routing packets rather than switching minutes, and charging a flat fee per port with no transaction charges. VPF claims to switch an equivalent telephone traffic of $230 \cdot 10^9$ minutes per year, having started in 2004 with only $2.5 \cdot 10^9$ minutes.[3]

Another player has been Infiniroute Networks Inc., founded in the second quarter of 2004 and acquired by Transaction Network Services (TNS) in March 2006, which now offers its services in 28 countries. Infiniroute routes VoIP calls over the public Internet with route diversity, charging carriers a flat fee per port.

[2] *This estimate counts calls twice (in and out), i.e., the actual volume of minutes is half this value.*

[3] *Notice that this traffic is almost entirely internal to the United States.*

Different interpretations of voice exchange exist, depending on the voice-switching technology, the protocol layer where the interconnection occurs, the ownership of the network where voice traffic is carried, and the business relationship between customers and the exchange.

XConnect (http://www.xconnect.net) was launched in March 2005 to interconnect VoIP-over-broadband providers. XConnect performs peering at layer 4, offering electronic numbering (ENUM) lookup[4] and interoperability services, to ensure that VoIP providers with different interpretations of the Session Initiation Protocol (SIP) stack or VoIP providers using other protocols such as H323, can interconnect. It also provides both settlement, without underwriting risk, and settlement-free minutes trading. XConnect provides services to over 400 VoIP operators in more than 35 countries.

Arena is a Europe-wide service of Interoute (http://www.interoute.com) consisting of a softswitch peering platform. It was launched in the second quarter of 2005 with a VoIP-switch partitioning service. Interoute provides many other data wholesale and business retail services with a network coverage of 85 cities in 22 countries across 54,000 km of fiber.

COMPARISON OF EXCHANGE MODELS

As just highlighted, different interpretations of voice exchange exist, depending on the voice-switching technology, the protocol layer where the interconnection occurs, the ownership of the network where voice traffic is carried, and the business relationship between customers and the exchange.

As to *voice-switching technology*, Arbinet provides TDM-circuit-switched interconnection natively; it has provided VoIP interconnection only since October 2004. To compete with newer players, Arbinet introduced a set of new commercial VoIP interconnection services that allow VoIP service providers to receive a per-minute termination fee for all successful calls from Arbinet's VoIP and TDM members to their VoIP customers. In contrast, all other providers are natively IP-based, with Infiniroute also handling TDM-IP interworking. VPF growth is mainly due to the migration of customers to VoIP, whereas the volume growth of Arbinet in 2007 (+14 percent) reflects the growth of mobile traffic (44 percent of the total Arbinet traffic in 2006) plus a share of VoIP growth (30 percent of the total Arbinet traffic in 2006). In addition, IP interconnection also allows interconnection for services other than VoIP.

Interconnection can take place at any layer of the open systems interconnection (OSI) stack: the higher the protocol layer, the higher the value in providing the transit service. Arbinet and Infiniroute route traditional voice and VoIP calls natively. Interoute routes VoIP calls. XConnect routes VoIP calls and manages address-translation requests. Stealth's VPF routes IP packets carrying voice.

Another important difference consists in the *ownership of the network* over which voice is carried: over the Internet (Infiniroute and XConnect) or over the enterprises' own IP networks (Arbinet, VPF, and Interoute). This factor makes a big difference in terms of expected quality and cost of service. Arbinet and Interoute also outsource voice infrastructures and operations, providing operators with dedicated end-to-end network resources.

In terms of *business relationships*, Arbinet routes calls, underwrites credit risk, and is a marketplace for anonymous-minutes trading. In contrast, all of the other exchanges leave interconnected operators to fix their own arrangements and do not underwrite risk.

EVOLUTION OF ROUTING MECHANISMS FOR INTERCONNECTION

In traditional TDM-circuit interconnection, a voice call traverses the network, passing through different switches, until it reaches the destination switch, as shown in Fig. 3a. Every switch along the path receives call-signaling messages and then processes and transfers them to the next one along the path. If the calling and called parties belong to networks of two different operators and the two networks are not directly interconnected, one or more transit networks are interposed. Every switch in each network (originating, transit, and destination) is involved in routing the call to the destination, processing signaling messages, and making routing decisions, based on predefined routing databases and call codes. The disadvantage of this mechanism is that calls are routed switch by switch, with a relevant call set-up overhead in terms of time and cost.

In a pure VoIP call, instead, after the conversion between called telephone number and corresponding IP address has been made, the two parties can talk to each other directly with no further involvement of intermediate switching elements, provided that an underlying IP network is between end points. This mechanism is usually referred to as *soft-switching* (Fig. 3b). When calling and called VoIP terminals belong to different VoIP providers, their soft-switching elements must exchange call signaling and routing information. If a transit network is involved, the intermediate soft-switch must route the call from the origin to the destination network. In this scenario, voice exchanges are transit networks that switch or soft-switch calls and play the man-in-the-middle role between the two interconnected parties, which have no visibility of each other's addresses and must rely on someone else to route the call. The main advantage of this mechanism is that call routing within the network domain of one operator occurs in one place only (the soft-switch), reducing the number of intelligent devices in the network and therefore, capital and operational costs.

A newer and common routing mechanism enables partners to share addresses and routing information so that they can directly route the call from origin to destination without the requirement of intermediate switching elements, even if they are not directly interconnected. This scenario is often referred to as *VoIP peering* (Fig. 3c). It is done through a shared, centralized address repository provided by the exchange, which can be exported to provider-specific servers. The voice exchange provides the registry, manages trading and settlement, and provides IP connectivity between the VoIP providers and the TDM-IP interworking facilities between TDM and VoIP providers. Recently, Arbinet introduced a service based on this approach, charging a fee for every successful query.

VoIP peering allows two network operators, interconnected along a chain of IP networks, to set up voice calls with no involvement of intermediate operators, which are confined to IP routing, thus reducing the complexity of the

[4] *ENUM is a standard protocol of the Internet Engineering Task Force (IETF), which uses the Internet Domain Name System (DNS) to translate ordinary telephone numbers into IP addressing schemes (e.g., SIP, H323, email, or uniform resource locators). The goal of the ENUM standard is to provide a single number to replace the multiple, underlying contact details for an individual's home, business, and cellular phones, as well as fax, email, etc.*

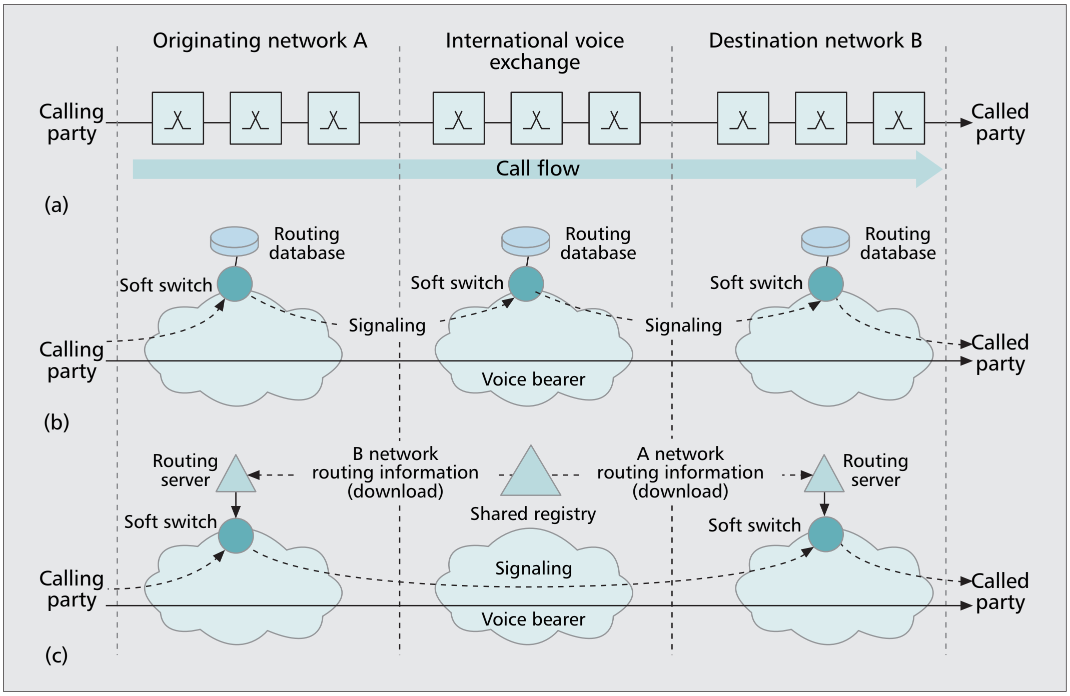

Figure 3. *Evolution of routing mechanisms for interconnection: a) TDM interconnection; b) VoIP interconnections; c) VoIP peering.*

As the convergence of services (including voice) over IP takes place, one might assume that VoIP interconnection will necessarily follow the B&K principle, mainly due to its simplicity and its success for the Internet. However, this conclusion should not be taken for granted.

interconnection model from both economical and technical viewpoints. The main disadvantage is that the full matrix of operators must exchange routing information, thus creating the business for VoIP-based voice exchanges.

The Evolution of Interconnection Commercial Models: Calling Party's Network Pays and Bill & Keep

Regardless of call routing, the future of voice exchanges depends on how commercial interconnection models evolve. Two opposite approaches are common: *Calling Party's Network Pays* (CPNP) and *Bill & Keep* (B&K).

The CPNP interconnection model is founded on the *Calling Party Pays* (CPP) billing regime on the retail side and assumes that the originating operator must pay the destination operator for routing the call to termination. The CPNP principle is the one behind the traditional minute-based settlement mechanisms, adopted by telephone operators with few exceptions (e.g., mobile operators in the United States).

In contrast, the B&K interconnection model assumes that nothing is due for calls originated by *A* and terminated on *B* between operators *A* and *B* and vice versa. The B&K principle is the one currently adopted in the Internet for IP traffic exchange between Internet service providers (ISPs). Its extension to voice traffic was proposed in the United States in 2000 [3]. The B&K model is still advocated today by its original promoter [4] and other economists, arguing that the CPNP principle is founded on the erroneous belief that the originator of a call is the *cost causer* and should then bear all the implied costs (as in CPP).

The opposite approach to CPP billing is called *Receiving Party Pays* (RPP). It dictates that because the receiving network has regulatory obligations to terminate calls without charge, operators must recover termination costs by charging its own customers for incoming calls. Recent studies [5] have confirmed that the RPP regime has a negative impact on mobile business due to the resistance of customers to pay for receiving calls.

The B&K interconnection model, which follows the RPP billing principle, is based on the assumption that both parties are *responsible for continuing a call*, no matter which party initiated it. Thus, it establishes that each party is responsible for the cost of its local access, and the calling party is responsible for the cost of carrying the call up to the destination central office. Operators are left free to choose whether to charge customers for the received calls or not.

As the convergence of services (including voice) over IP takes place, one might assume that VoIP interconnection will necessarily follow the B&K principle, mainly due to its simplicity and its success for the Internet [6, 7]. However, as argued in the next section, this conclusion should not be taken for granted.

Value of Call Termination: Choosing the Interconnection Commercial Model

The assumption that the operating cost of switching one minute in a VoIP network is less than in a TDM network is widely accepted (although not always verified). The idea that this cost equals zero just because the switching is through IP is a popular misunderstanding, perhaps due

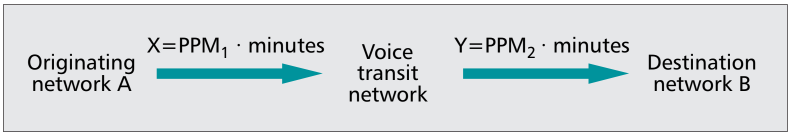

Figure 4. *Traditional business scenario, in which voice traffic is carried through a transit network.*

to a misinterpretation of the B&K principle. For B&K, the cost of a call is not null itself, but rather, it should be shared by the two parties instead of being paid by the originating side only.

If the value of exchanged traffic between two operators is balanced, a free peering arrangement can be chosen with no impact on profits (but only on revenues). If this value is unbalanced, then some sort of compensation must be established. Thus, the real issues are to assess the value of mutually exchanged traffic to determine if it is balanced and how to set the unbalance, if any.

Value of Interconnection

Note that the main value of a communication network is its capability to put its users in communication: the larger the network, the greater the value provided. The opportunity for operators to interconnect must be seen in terms of the increase in the number of reachable users. Smaller operators benefit more from interconnection with larger ones, as is obvious.

When telecommunications markets were liberalized worldwide, incumbents were forced to provide interconnection with new entrants. Tariffs for terminating calls to and from the incumbent network were fixed by local regulators, typically in an unbalanced way (i.e., calls terminated on the incumbent were priced lower), based on the consideration that new entrants must pay back new infrastructures, while incumbents operated fully amortized networks.

Therefore, whereas in a non-regulated market, considerations about operator size suggest that the value of interconnection is higher for smaller players, in a regulated market, this size/value equation might be reversed. This demonstrates that value balance must not be assumed a priori.

Cost Assessment

Retail and wholesale prices are closely related: an operator would never choose a retail price for a call that was lower than the regulated cost to terminate the call on the destination network (e.g., fixed-to-mobile). In other words, the commercial definition of interconnection depends on retail economics (and vice versa): the choice between a balanced/free or unbalanced/settled interconnection model largely depends on the specific service, the provider, and the country regulations on the two retail sides.

In addition, no commercial interconnection model is feasible without considering the actual utilization of network resources that the originating operator is borrowing from the destination operator. Resources can be divided into four different kinds: *interconnection resources* (i.e., physical and logical ports dedicated to interconnection), *transport resources* (i.e., transmission links, routers, switches, and gateways from the interconnection point to the destination), *service resources* (e.g., numbers or addresses, signaling gateways, soft-switches, media servers, applications, etc.), and *access resources* to reach the destination terminal.

All these components contribute to the overall cost for determining the lower bound of the interconnection price. Clearly, each term has a different weight in the total cost. We note that whereas the interconnection value depends on the cost of used resources, the B&K principle, on the contrary, is an incentive for operators to handover traffic onto the networks of other operators as early as possible ("hot potato" effect), without any recognition of the termination services rendered.

Quality of Service

Finally, the price paid for call termination also should take into account the quality of service provided by the destination network (e.g., in terms of resource-seizure probability, voice quality, reliability, etc.). Also in this case, assuming symmetry at the interconnection point is far from reality. For example, in Arbinet, the probability of seizing the call on a destination route impacts the unitary price of call termination on that route. It is almost certain that future commercial interconnection models will take into account quality and price differentiation, whatever the service will be.

The Commercial Model for Interconnection

The B&K principle is probably too simplistic for services other than mere best-effort Internet connectivity. Interconnection agreements, instead, will depend on country-specific market scenarios and regulation, as well as on service-specific retail economics, cost considerations, and quality issues. In addition, different (albeit correlated) models can be envisaged depending on where interconnection takes place: in the access network, in the core national network, or at the international level. In any case, it must be recalled that B&K is promoted as a *default* regime for all those cases in which two operators "cannot agree on the terms of interconnection" [3], whereas national regulators must establish rules to guarantee fairness [7].

The Evolution of the Interconnection Charging Models

Given that an exchanged-traffic value should be mutually recognized, now the issue is how unbalances should be settled. If *flat-fee charging* is applied on retail (e.g., VoIP providers), it is likely that flat-fee is adopted also at interconnection. Yet, while voice is more and more commoditized, the same cannot always be assumed for other services, like multimedia messaging [8], video communications, and content-based services in general.

In addition, in flat-fee interconnection schemes, the fee must be established based on the predicted number of service units traversing the border. Between two public-switched telephone network (PSTN) operators, units are minutes. Between two ISPs, units are IP packets. Between two SIP-based VoIP providers, units are sessions (or their duration). Between two mobile telephone operators, units can also be text messages.

Instead of counting voice minutes, packets, sessions, or messages and mutually billing the amount of exchanged units, a predefined volume can be assumed, based on previous observations. If the actual volume of exchanged traffic changes, then fees should be adjusted as well. Thus, the adoption of a flat-fee model still requires resource usage monitoring tools on both sides of the interconnection.

Finally, flat fees are usually based on interconnection port capacity at the underlying network layer, depending on the specific service. For example, in PSTN interconnection, flat fees can be established based on the number of E1 circuits, while in VoIP interconnection, fees can be established based on the Ethernet bandwidth allocated at the peering point. This simple but effective approach can be followed as long as network capacity is exclusively reserved to each service at the interconnection point, and the correlation between network capacity and price is service-specific. In other words, interconnection must be service-aware.

In conclusion, flat-fee models are not necessarily the right choice for all services, do not eliminate the requirement for resource-usage monitoring capabilities, and do require separate treatment for IP traffic and the distinct correlation between network capacity and fee for different services. When a B&K model does not apply, some sort of compensation for value unbalance between operators should be established, either usage-based or flat-fee.

Future Business Scenarios of Voice Exchanges

We now discuss how the evolution of interconnection routing mechanisms will impact the business of voice exchanges. First, we must distinguish between the cases in which voice traffic is carried through a transit network or an exchange.

In the traditional business scenario where voice traffic is carried through a *transit network* (Fig. 4), the originating network A pays call termination (X) to the transit network, which then pays a smaller amount ($Y < X$) to the destination network B. Amounts X and Y are given by the price-per-minute (PPM) agreed, multiplied by the number of minutes carried. This is the scenario that originally created the business of voice exchanges — to save on the transit cost.

Alternatively, voice traffic can be carried through an international voice exchange, as shown in Fig. 5. In the first scenario of this kind (Fig. 5a), voice traffic is circuit-switched in TDM (*TDM interconnection*). The buyer (i.e., the originating operator A) pays the amount $PPM \cdot minutes$ for call termination to the seller (i.e., the destination operator B), with the optional settlement mediation by the voice exchange. Moreover, buyer and seller also pay the exchange a per-minute fee, related to the price-per-minute agreed in the transaction. Finally, the seller may also pay an additional fee for the credit risk underwritten by the exchange. On the average, voice exchanges make the cost of taking the call from A to B lower than in the transit-network scenario of Fig. 4 by leveraging the arbitrage made possible by the high costs of long-distance voice switching. This is what Arbinet did first (1996) and is still doing for parties interconnected with TDM circuit-switched networks.

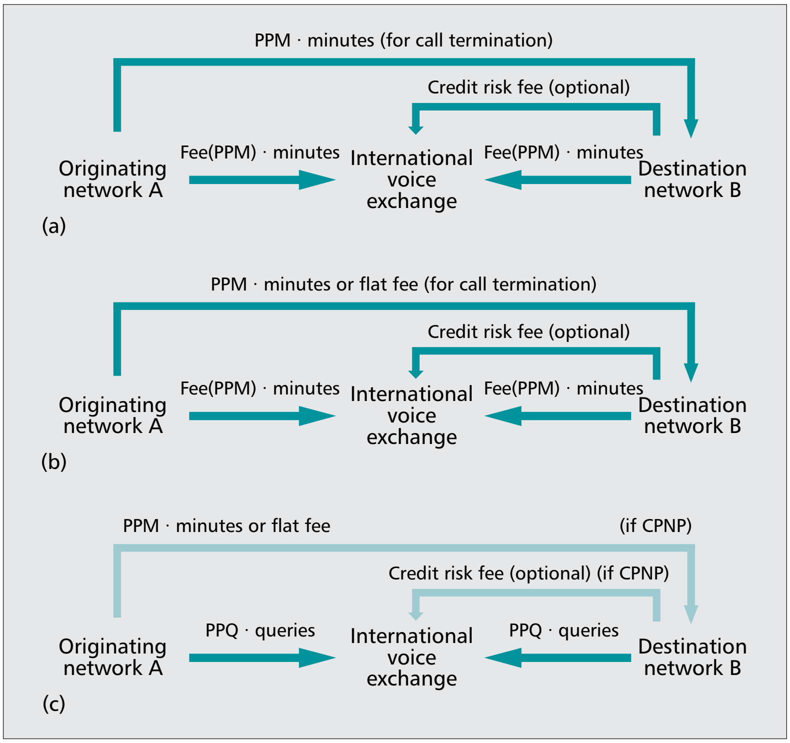

Figure 5. *Alternative business scenarios, in which voice traffic is carried through a voice exchange: a) TDM interconnection; b) VoIP interconnection; c) VoIP peering (CPNP or B&K).*

In principle, a very similar business scenario also can be envisaged when one or both operators require *VoIP interconnection* (Fig. 5b). This may be the preferred scenario by incumbent and alternative — both fixed and mobile — operators who have TDM circuit-switched networks gradually migrating to VoIP, while preserving the wholesale revenue coming from call termination. Flat-fee instead of per-volume charging for call termination from A to B can be chosen if retail billing is flat on both sides. This scenario is still not applied widely. Many telephone operators are sticking to a TDM-interconnection model and still see VoIP interconnection as a threat to their termination revenues. For instance, in Italy, FASTWEB has a full VoIP network, but its interconnection to Telecom Italia and other operators is still TDM-based.

In the case of *VoIP peering* between operators (Fig. 5c), two possible business scenarios can be further envisaged depending on the commercial model agreed upon by parties A and B, namely, CPNP or B&K. In both cases, the voice exchange

As these operators gradually migrate customers to VoIP, they will initially transfer the existing termination scheme over their upgraded networks to preserve their market position. It is not easy to predict the way that interconnection regimes will evolve.

receives a fee for successful registry queries, based on a predefined price-per-query (PPQ).

If CPNP is chosen, typically by fixed and mobile operators on TDM circuit-switched networks, the buyer A pays the seller B for the call termination, with the optional settlement mediation by the voice exchange. Charging between A and B can be per-volume or flat-fee, depending on the specific agreement between the two parties. Note that the exchange does not charge parties based on call duration because it has no switching resources allocated to the call during its extent. Yet, exchange charges are still per-volume but in this case, volume consists of *queries* instead of *minutes*. Optionally, exchange charges also might be flat-fee, based on an estimated number of queries.

Instead, if B&K is chosen, no call termination is paid by A to B, and no mediation by the voice exchange is required: the exchange is merely playing the role of *route server*. Also in this case, registry queries can be charged either per-volume (i.e., $PPQ \cdot queries$) or flat-fee.

VoIP peering is the model of voice exchange, for example, implemented by XConnect. Interconnection becomes more a matter of exchanging routing information than routing calls along a chain of interconnected networks, thus yielding a reduction of overall complexity and cost.

Conclusions

In this article, trends are highlighted that are reshaping the business of the ILD voice market. In this crowded and unregulated market, voice exchanges have played a major role since the mid-nineties, even more so after the introduction of VoIP.

Various existing models of voice exchanges are outlined. The evolution of interconnection routing mechanisms and the consequent role of exchanges are summarized. The CPNP and B&K interconnection commercial models are introduced, including a discussion of the issues for their applicability. Flat-fee charging models for interconnection between operators are discussed. Finally, the impact of these models on future business scenarios of voice exchanges is analyzed.

Our description points out that the common belief that VoIP will eliminate the CPNP interconnection model, in favor of B&K, should not be taken for granted. The applicability of the B&K principle in operator interconnection, as well as pure flat-fee schemes, typically is well suited for best-effort services like Internet connectivity, but must be assessed against existing regulations, current charging principles, new services, new business models, and above all, market dynamics.

It can certainly be presumed that B&K, or some flat-fee or semi-flat-fee charging schemes, eventually will be adopted by IP-based providers for voice services. Yet, it is unlikely that a similar approach will be easily followed by incumbent and alternative operators, either fixed or mobile, for their TDM circuit-switched networks.[5]

As these operators gradually migrate customers to VoIP, they will initially transfer the existing termination scheme over their upgraded networks to preserve their market position. It is not easy to predict the way that interconnection regimes will evolve. However, B&K does not seem to fit the complexity of most current and future situations. Voice exchanges must continue to adapt their businesses to existing and emerging models.

[5] *See also responses from British Telecom, France Telecom, Deutsche Telekom, Vodafone, and the GSM Association to ECC Consultation [6].*

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Biographies

STEFANO BREGNI [M'93, SM'99] (bregni@elet.polimi.it) is an associate professor at Politecnico di Milano, Italy. Since 2004 he has been a Distinguished Lecturer of the IEEE Communications Society. He is Director of Education and Chair of the TAOS Technical Committee of the IEEE Communications Society. He is Symposia Chair of GLOBECOM '09 and has been a Symposium Chair for seven other ICC/GLOBECOM conferences. He is the editor of the *Global Communications Newsletter*. He is the author of the book *Synchronization of Digital Telecommunications Networks* (Wiley, 2002).

GIACOMO BRUZZI (bruzzi@ictc.it) received his Dott. Ing. degree in electronic engineering from Politecnico di Milano in 1990. He has been CTO at ICT Consulting, Italy, since 2000. He currently works as a strategic/technical advisor for telecommunications operators and business companies. He began his career at Italtel, working on the first development of an ATM switch in Italy. In 1998 he joined Alcatel (Belgium and United States) as product manager for the IP products division.

MAURIZIO DECINA [M'73, SM'85, F'87] (decina@elet.polimi.it) has been a professor at Politecnico di Milano, Italy, since 1988. He was executive cirector of R&D at Italtel, Milano. He founded the CEFRIEL research and education consortium. He was a member of the board of Telecom Italia, Italtel, and Tiscali. He was President of the IEEE Communications Society (1994–1995). He received three IEEE awards: International Communications (1997), Third Millennium Medal (2000), and ComSoc/KICS Global Service (2008).