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July 23, 2019

Ms. Marlene H. Dortch
Office of the Secretary
Federal Communications Commission
445 Twelfth Street, NW
Washington, D.C., 20554

Re: WC Docket No. 19-44, Petition for Declaratory Ruling in Response to Primary Jurisdiction Referral, *Autauga County Emergency Management Communication District et al. v. BellSouth Telecommunications, LLC*, No. 2:15-cv-00765-SGC (N.D. Ala.)

Dear Secretary Dortch:

As the Alabama 911 Districts' prior *Ex Parte* filings indicated, Brannon Buck and Christopher Driver of Badham & Buck; Roger Schneider of Expert Discovery; and Jim Baller and Sean Stokes of Baller Stokes & Lide participated in several meetings with members of the Commission staff on June 18, 2019. In one of those meetings, a member of Commission staff in the Wireline Competition Bureau raised several questions and requested additional information on certain issues, including whether the Commission should address the Interconnected Voice over Internet Protocol (IVoIP) definitional question raised in the Districts' and BellSouth's petitions and the relevance of a finding that 47 U.S.C. § 615a-1 does not preempt state authority to impose 911 fees on service other than IVoIP. This filing responds to those questions and additionally addresses other issues raised in recent filings by BellSouth and other commenters.[1]

I. The Commission Should Address the IVoIP Question Referred by the District Court.

The Commission should address the IVoIP definitional question in this proceeding because the District Court unquestionably referred that issue to the Commission. In fact, BellSouth expressly requested a primary jurisdiction referral based on the IVoIP definition. Now, over a year after the referral order, BellSouth is asking the Commission *not* to decide a key issue on which it sought guidance. This recent change from BellSouth's position is inconsistent with the text of the District Court's order granting BellSouth's motion seeking a primary jurisdiction referral. The Court expressly discussed the IVoIP definitional issue at length, granted the motion, and instructed the parties to begin a proceeding with the Commission. The parties were therefore required to seek a declaration on the IVoIP issue, and both did so in their petitions. Similarly, if the Commission

[1] For reasons that the Districts do not understand, BellSouth is named in some filings in this proceeding and AT&T is named in others. For the purpose of this submission, unless the context indicates otherwise, the Districts will use BellSouth and AT&T interchangeably.

intends to address the issues on which the District Court sought guidance, it should address the IVoIP definitional issue.

Furthermore, BellSouth's argument that the Commission can sidestep the IVoIP definitional issue by simply holding that 47 U.S.C. § 615a-1 precludes states from discriminating against IVoIP providers defies common sense. How can the parties and the District Court compare IVoIP providers to traditional service providers to determine whether undue discrimination exists without a meaningful definition of "IVoIP" in the circumstances at issue before the District Court?

The Commission should also address the IVoIP definitional issue to avoid a far more time-consuming, burdensome, and costly process. If the Commission only addresses the preemption issue and does not address the IVoIP definitional issue, then whichever party is adversely affected could request that a court—either a federal court of appeals or the District Court—reverse the Commission. If the Districts are ultimately successful, then they will eventually be back in front of the District Court with no guidance on the IVoIP issue—which is just where the parties were at the time of the primary jurisdiction referral. At that point, the District Court will be faced with the decision whether to interpret the term "IVoIP" itself without Commission guidance, which it previously declined to do, or to refer the matter to the Commission again. This piecemeal approach will drastically extend a delay that is already 16 months long. The parties should be allowed to move their case through the judicial system without undue delay. Regardless of the Commission's views on Section 615a-1 preemption, it should address the primary issue referred by the District Court.

II. 47 U.S.C. § 615a-1 Does Not Preempt State Authority to Impose 911 Fees on Services Similar to VoIP.

The Commission should declare that 47 U.S.C. § 615a-1 does *not* preempt a state from imposing 911 fees on services other than IVoIP, commercial mobile services, or telecommunications services. The relevance of this issue is rooted in the Alabama statute at issue in this proceeding—the Emergency Telephone Service Act, Ala. Code § 11-98-1, *et seq.* (effective until October 1, 2013) ("ETSA"). The ETSA includes a provision that imposes 911 fees on "VoIP or similar service:"

> The emergency communication district fee authorized and levied in each district pursuant to Section 11-98-5 shall apply to all wired telephone service utilized within the district, including such service provided through Voice-Over-Internet Protocol (VoIP) *or other similar technology*. It shall be the duty of each provider of VoIP *or similar service* to collect the fee for each 10-digit access number assigned to the user and to remit such fee as provided in Section 11-98-5.[2]

[2] Ala. Code § 11-98-5.1 (*emphasis* added).

This provision was added to the ETSA in May 2005, so it predates the Commission's *E911 IP-Enabled Order*.[3] It also predates 47 U.S.C. § 615a-1, which became law in 2008.

Before the Alabama legislature acted in 2005, 911 fees in Alabama were generally based on channelized access lines. In light of the emergence of VoIP services that allowed access to 911, the ETSA imposed 911 fee obligations on those services to ensure that the emergency management operations they affected would be adequately funded. As the Commission recognized in its *E911 IP-Enabled Order*: "[W]e recognize that while some state laws today may already require 911 funding contributions from providers of interconnected VoIP, interconnected VoIP providers may not be covered by existing state 911 funding mechanisms in other states."[4]

Because Section 5.1 of the ETSA was forward looking and predated both the *E-911 IP-Enabled Order* and 47 U.S.C. § 615a-1, the state of Alabama could not rely on the Commission's definitions and categorizations of VoIP services. In the absence of Commission guidance, and in an effort to ensure that all services utilizing and impacting E-911 capabilities and resources were subject to contribution obligations, the ETSA adopted a broad and inclusive category of services subject to the statute —"Voice-Over-Internet Protocol (VoIP) or other similar technology…VOIP or similar service."[5] By imposing fees on *similar* services, the ETSA was again forward-looking. It prevented a potential gap in coverage through which providers of voice services that were neither strictly VoIP nor strictly traditional wireline could avoid billing 911 fees.[6] In other words, the Alabama Legislature could not predict what VoIP-like services might emerge after the passage of Section 5.1 of the ETSA, so it ensured that new services that shared some characteristics of VoIP and took advantage of the 911 system faced the same 911 fee requirements as VoIP, wireline, and wireless services.

In the District Court case underlying this proceeding, this particular type of service—service that is neither strictly VoIP nor strictly traditional wireline—is potentially at issue.[7] Specifically, in their initial petitions, the parties included scenarios illustrating various potential service configurations. Scenario 3b depicts a service that transmits voice over the last mile in Ethernet packets. That service is not IVoIP because it uses Ethernet packets rather than IP packets, and it is not traditional telephone service because it uses packet-switching rather than circuit-switching. As BellSouth notes in its petition, the "Commission has repeatedly recognized that the core distinction between traditional telephone service and VoIP is that traditional telephone service

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[3] *In the Matter of E911 Requirements for IP-Enabled Service Providers*, WC Docket No. 05-196, *Report and Order* (*"E-911 IP-Enabled Order"*), (rel. June 3, 2005).
[4] *E-911 IP-Enabled Order* at ¶ 52.
[5] Ala. Code § 11-98-5.1.
[6] Importantly, the ETSA does not regulate VoIP or similar services or impose requirements outside of 911 fees.
[7] This service is *potentially* at issue because BellSouth has not yet responded to discovery showing what services it actually provided during the relevant timeframe.

transmits voice communications *over the circuit-switched network*, while VoIP instead uses IP technology *and packet switching*."[8]

The service depicted in Scenario 3b interconnects with the PSTN, and it allows subscribers to call 911. As a result, Scenario 3b places a burden on local PSAPs and should be assessed 911 fees. Yet, such service falls outside the definitions of both IVoIP and traditional switched-circuit services. Section 5.1 of the ETSA was drafted to address this potential loophole by imposing 911 fee obligations on services like those in Scenario 3b that do not qualify as VoIP but are similar to VoIP and impact the 911 PSAP system. The Districts have alleged in the lawsuit that BellSouth has failed to bill the correct number of 911 fees—one per every 10-digit access number up to a cap of 100 per person per location—for its VoIP *and similar services*.

The Districts and BellSouth are *not* asking the Commission to interpret "similar service" or "similar technology." Nor are they asking the Commission to declare that Scenario 3b is similar to VoIP. The interpretation and application of the ETSA are for the District Court.

Rather, the Districts request that the Commission, if it addresses preemption, find that 47 U.S.C. § 615a-1 does *not* preempt states' authority to impose 911 fees on services other than commercial mobile services, interconnected VoIP, and telecommunications service—the three services expressly identified in Section 615a-1. The Districts believe that it is very important for the Commission to do so because BellSouth has erroneously argued that Section 615a-1 allows states to impose 911 fees *only* on IVoIP services. For example, BellSouth argued to the District Court (with our emphasis added):

> … Plaintiffs are wrong to assert (at 9-10) that Congress and the FCC left states with wide authority ... to expand the category of VoIP services subject to state 911 charges. *Under federal law, state authority to adopt statutes imposing 911 charges on VoIP services is limited to those services that qualify as interconnected VoIP under 47 C.F.R. § 9.3*.[9]

Other telephone service providers—specifically, Windstream—have also argued in similar lawsuits that Section 615a-1 prohibits states from imposing 911 fees on any services other than IVoIP.

In its filings in this case, BellSouth has now abandoned this argument and conceded that that states have authority to impose 911 fees that does not derive from Section 615a-1.[10] The Commission should accept BellSouth's concession and put this matter to rest, once and for all.

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[8] AT&T Comments in Opposition to the Alabama 911 Districts' Petition for Declaratory Ruling, March 28, 2019, at 1 (*emphasis* added).

[9] *Autauga Cty. Emergency Mgmt. Commc'n Dist. v. BellSouth Telecomms., LLC*, No. 2:15-cv-00765-SGC (N.D. Ala.). Dkt. 42 at 8.

[10] AT&T Comments in Opposition to the Alabama 911 Districts' Petition for Declaratory Ruling, March 28, 2019, at 19–20.

As the Districts showed in their Petition (at 28–35), Section 615a-1 codified the Commission's determination in its *E-911 IP-Enabled Order* that all providers of IVoIP must make 911 service available to their customers. Section 615a-1 does not purport to limit the authority that state and local governments *already* had to require service providers to make 911 service available to their customers through IVoIP or other means and to charge 911 fees on such services. In fact, in its *Order*, issued well before Congress enacted Section 615a-1, the Commission recognized that state and local governments already had ample authority to impose 911 requirements and fees and that many of them had already exercised such authority in imposing 911 obligations on VoIP service providers:

> The availability of this critical service is due largely to the efforts of state and local authorities and telecommunications carriers, who have used the 911 abbreviated dialing code to provide access to increasingly advanced and effective emergency service capabilities. *Indeed, absent appropriate action by, and funding for, states and localities, there can be no effective 911 service. Responsibility for establishing and designating* PSAPs or appropriate default answering points, purchasing customer premises equipment (CPE), retaining and training PSAP personnel, purchasing 911 network services, *and implementing a cost recovery mechanism to fund all of the foregoing, among other things, falls squarely on the shoulders of states and localities*.
>
> …
>
> We believe that the requirements we establish today will significantly expand and improve interconnected VoIP 911 service while substantially reducing the threat to 911 funding that some VoIP services currently pose. *First, we recognize that while some state laws today may already require 911 funding contributions from providers of interconnected VoIP*, interconnected VoIP providers may not be covered by existing state 911 funding mechanisms in other states.[11]

Therefore, if the Commission addresses preemption, it should reaffirm two basic points: (1) state authority to impose 911 fees is not derived from Section 615a-1; and (2) as a result, states have the authority to impose 911 fees on services other than those expressly identified in Section 615a-1. This finding would affirm the Districts' authority to enforce the ETSA's imposition of 911 fees on service "similar" to VoIP.

III. VoIP Service and Traditional Service Are Not Similarly Situated.

BellSouth argues that the Commission should find that any state statute that discriminates against IVoIP by charging customers more than they would pay for similar traditional service is preempted by Section 615a-1. BellSouth's preemption argument is based on the premise that IVoIP services and traditional telecommunications services are similarly-situated and that, to avoid discrimination, it is necessary to subject IVoIP service providers to the same technological or legal constraints that apply to traditional service providers. This premise is false. VoIP and traditional services are fundamentally different. As a result, assessing 911 fees on a spurious similarly-

[11] *E-911 IP-Enabled Order* at ¶¶ 7, 52 (emphasis added).

situated rationale—i.e., by call capacity or concurrent call capacity as suggested by several of the telecom commenters—is not only impractical, but it would also lead to absurd outcomes.

As the Districts noted in their reply comments in this proceeding, IVoIP service offers numerous capabilities that make it more flexible, scalable, and efficient for customers than traditional telephone services.[12] For example, VoIP can offer a burstable concurrent call limit under which a subscriber may contract for a certain number of simultaneous calls but be able to make more simultaneous calls than the maximum in the contract. The provider may or may not charge subscribers for this capability, but the 911 system must bear the costs of being prepared for the additional calls. For example, Verizon offers a service through which a "Customer can exceed (or 'burst') its simultaneous calling capacity if, for example, it experiences an unplanned burst of inbound/outbound voice calls."[13] Verizon provides the following chart to show its VoIP product's bursting capacity:

Simultaneous Calling Capacity		
BEST+ Tier	**Per Enterprise***	**Maximum Additional Simultaneous Calling Capacity**
1	200 – 399	+ 50
2	400 – 799	+100
3	800 – 1,199	+ 200
4	1,200 – 1,599	+ 300
5	1,600+	+ 400
*Customer may purchase at its Per Enterprise level or below. For example, if Customer purchases a Simultaneous Calling Capacity of 1,000 calls, it is in BEST+ Tier No. 3. It can purchase the Maximum Additional Simultaneous Calling Capacity for Tier Nos. 3, 2, or 1. It cannot purchase at Tier Nos. 4 or 5 (unless it subsequently purchases additional Simultaneous Calling Capacity to advance into either of those Tiers).		

[14]

According to this chart, a Verizon VoIP subscriber can contract for a simultaneous call capacity of 1,000, but in reality, it could make 1,200 simultaneous calls at any time.

Traditional, circuit-switched voice services do not have this bursting capability. For example, a traditional PRI has the capacity to make 23 simultaneous calls. This is a strict physical limit, unlike VoIP products that can have tens to hundreds more simultaneous calls above the contracted limit.

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[12] The many additional benefits that customers receive from the services at issue here distinguish these services from the traditional services at issue in the *IP-in-the-Middle* case, *In the Matter of Petition for Declaratory Ruling that AT&T's Phone-to-Phone IP Telephony Services are Exempt from Access Charges*, *Order*, 19 FCC Rcd. 7457, 2004 WL 856557 (rel. April 21, 2004).
[13] *See* Ex. A p. 10- Sec. 5.1.2.4. Exhibit A is an excerpt from an Exhibit A of the Districts' Reply Comment.
[14] *Id.*

In addition to bursting capabilities, VoIP offers the ability to share simultaneous call capacity among segments of an enterprise with multiple locations—including locations in different cities, counties, or states. AT&T offers this type of service:

> Bursting and sharing is functionality that allows for calls that go beyond the maximum number allowed on the trunk to borrow call capacity from another group trunk group within your enterprise. The maximum group trunk group burst can't exceed 20% of a group trunk group's maximum concurrent call limit.[15]

Verizon similarly offers the ability to share concurrent call capacity with other locations within a business subscriber's enterprise:[16]

> **Burstable Enterprise Shared Trunks (BEST).** Customer's VoIP sites that are provisioned with BEST will be able to share the total simultaneous calling capacity purchased by Customer across its enterprise on a regional basis. Thus, simultaneous call units within a region contribute to the total available concurrent call capacity only within that region. Concurrent call pools cannot be regionally shared between the U.S./Canada, Europe, and Asia-Pac regions. BEST applies to enterprises in which all locations are on a metered or tiered pricing model. Simultaneous calling capacity can be shared between locations receiving both Local and LD VoIP service, and between locations receiving only LD service, but not across those two kinds of locations.

This BEST service specifically allows subscribers to share simultaneous calling capacity among multiple locations. In other words, a business could share a simultaneous call capacity of 500 calls among three different locations—with each location having the ability to make 500 simultaneous calls at any particular time.[17]

AT&T, Verizon, and CenturyLink suggest in their joint *ex parte* filing of July 11, 2019 that traditional TDM services also have the ability to "share their purchased outbound calling capacity to the PSTN." They are not clear on the specifics of such sharing, but they reference a Cisco product with a feature that allows a business with a "large number of locations" to "decouple the location of the PSTN gateway" from the locations of the employees making telephone calls. Presumably, this scenario is based on multiple locations with a TDM PRI between a location and the headquarters where the PSTN gateway is located. While the different locations share connections to the PSTN in the sense that all locations' calls go through the main gateway, the individual locations cannot share the call capacity of the entire enterprise. Specifically, each location with a TDM PRI leaving its premises as part of a Cisco local route group is limited to 23

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[15] *See* Ex. B at 196. Exhibit B is an excerpt from Exhibit C of the Districts' Reply Comments of April 12, 2019. A group of Telecom Commenters identify this AT&T service in an *Ex Parte* Notice and state that it can provide a service that has a call capacity of 23 simultaneous calls. This description is incomplete. According to AT&T's own marketing materials, IP Flexible Reach can have a burstable and shareable call capacity—which is a far cry from the over-simplified description in the *Ex Parte* Notice. June 7, 2019 Joint *Ex Parte* Notice.

[16] *See* Ex. A at p. 1, Sec.1.2.2.

[17] Obviously, in a shared environment, one location's ability to use the full 500 concurrent call capacity would be reduced by the number of calls, if any, being placed at the same time by the other locations.

simultaneous calls per PRI—the location cannot utilize the aggregate call capacity of the entire enterprise. IVoIP, on the other hand, imposes no such location-specific limitation. With IVoIP, each location has the call capacity of the entire enterprise.

Some VoIP systems can also be configured to allow 911 calls even if a subscriber is using all of its simultaneous call capacity.[18] Specifically, SIPTRUNK—a VoIP provider—advertises this particular capability: "Even when you're using all of concurrent calling capacity some SIP trunking providers will give priority to a 911 or e911 call and let the call go through."[19] Similarly, Cisco provides guidance to its product users on methods to ensure 911 calls are not blocked in "all trunks busy" situations.[20] Cisco notes that "[i]t is highly desirable to protect 911 calls from 'all trunks busy' situations" and that "[i]f a 911 call needs to be connected, it should be allowed to proceed even if other types of calls are blocked due to lack of trunking resources."[21]

In other words, VoIP subscribers can have the ability to make 911 calls even if other calls would be blocked by a provider-imposed simultaneous call cap. This ability, as Cisco observed, is "highly desirable" and is consistent with the general policy that 911 calls should not be blocked and should be connected to the appropriate PSAP.[22]

VoIP's ability to have burstable and shareable simultaneous call capacities, along with the ability to prioritize 911 calls, demonstrate that BellSouth's favored outcome—requiring the total number of 911 fees assessed on IVoIP and traditional service to be the same, i.e., by simultaneous call capacity—would be unworkable in many cases. The impracticality of this approach is rooted in the false premise that VoIP and traditional service are similarly situated. For instance, what is the traditional telephone service equivalent of a VoIP customer with a general simultaneous call capacity of 1,000 but the ability to burst to 1,200 calls? What is the traditional telephone service equivalent of a VoIP customer with a shared pool of 1,000 simultaneous calls across three locations? What is the traditional telephone service equivalent of a VoIP customer with contractual limit of 100 simultaneous calls but the ability to make 911 calls above the 100 simultaneous call capacity? Simply put, no traditional telephone service equivalent exists. Traditional, circuit-switched telecommunications services cannot *burst*, and they cannot *share* concurrent calls beyond a particular location's PRI-imposed physical limitation.[23] The enhanced capabilities of VoIP negate any direct simultaneous call capacity comparisons between traditional telephone services and VoIP services.

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[18] *See* Ex. C, Cisco, *Cisco Collaboration System 11.x SRND*, at 15-17 ("Cisco"); Ex. D, *How SIP Trunks Handle 911 Calls*, SIPTRUNK (January 13, 2016), https://www.siptrunk.com/2016/01/sip-trunking-e911/ ("SIPTRUNK 911").
[19] Ex. D, SIPTRUNK 911.
[20] Ex. C, Cisco at 15-17.
[21] *Id.*
[22] In its recent order on robocalling, the Commission emphasized that voice providers "should not block emergency calls and the Commission's rules prohibit voice service providers from blocking emergency calls to 911." *In the Matter of Advanced Methods to Target and Eliminate Unlawful Robocalls*, WC Docket No. 17-97, *Declaratory Ruling and Third Further Notice of Proposed Rulemaking*, (rel. June 7, 2019).
[23] Specifically, traditional service cannot *share* the total call capacity of an entire enterprise.

In a joint *ex parte* submission filed on July 11, 2019, AT&T, CenturyLink, and Verizon argued that "providers sell VoIP services that provide business customers the ability to make a limited number of simultaneous calls to the PSTN (including 911) . . . ." This statement may or may not reflect the providers' *contracting* practices, but it does not address how they actually handle 911 calls, given their customers' ability to burst, pool, and take other actions that could exceed contractual limitations.[24] Furthermore, the statement does not necessarily reflect the practices of other telephone service providers. The following marketing materials from VoIP providers demonstrate this point clearly:

- Vonage: "Vonage Business Plus plans allow direct SIP trunking from your VoIP Gateway, Asterisk/TrixBox, or other IP PBX directly to Vonage with **unlimited simultaneous incoming and outgoing calls**. . . . Vonage Business Plus are pay-per-minute plans that do not limit the number of simultaneous calls incoming or outgoing to your SIP trunk."[25]

- Twilio: "Concurrency refers to how many simultaneous calls can occur at same time. These traditional providers require you to plan for your highest peak traffic days, determine the highest number of concurrent calls that you may need to support, and pay for the number of 'channels' that support that traffic for the duration of your contract term. Twilio, simply put, doesn't require any upfront capacity planning—we allow you to scale as needed, and you won't be charged for concurrency."[26]

- Telzio: "Telzio support **unlimited concurrent calls**, which means all your employees can make calls using a single Telzio phone number without callers experiencing a busy signal. With Telzio, there is no need for rollover lines."[27]

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[24] Because the Districts have not received discovery from BellSouth in the underlying litigation, any representation by BellSouth that it always limits simultaneous calls (and did so during the time period in question) has not been verified by documents or sworn testimony. The Commission should not base its decisions on unverified representations and disputed factual assertions.

[25] Ex. E, at p. 1 (emphasis added), *available at* https://www.voip-info.org/vonage-business-plus/. Vonage's description of its VoIP offering reveals the speciousness of the statement by AT&T, CenturyLink, and Verizon that simultaneous call "limits are necessary so that providers can price their services appropriately." Vonage charges its subscribers on a "pay-per-minute" model, not the number of simultaneous calls.

[26] Ex. F, at p. 2, *available at* https://www.twilio.com/blog/2017/10/sip-trunking-termination-calling-more-flexible-master-cps.html.

[27] Ex. G, at p. 1 (emphasis added), *available at* https://telzio.com/support/making-outbound-calls. Telzio's product description notes that subscribes can make unlimited telephone calls with a single telephone number. This type of configuration demonstrates that BellSouth's repeated assertions that (1) subscribers have more telephone numbers than number of simultaneous calls and (2) imposing 911 fees on VoIP telephone numbers will lead to VoIP subscribers paying more 911 fees than traditional subscribers are not always accurate. Specifically, a Telzio subscriber can seemingly have unlimited simultaneous calls but only one telephone number. BellSouth's assertions are yet another attempt to create a factual paradigm that ignores the broader world of IVoIP and, instead, focus on the service offerings of a select few providers.

- Mitel: "With Mitel SIP, your technology can grow with your business. **Call capacity can be increased on-demand to support unlimited concurrent calls and fluctuations in call volumes**, while ensuring you never pay for more than you need."[28]

In short, while a contract-based standard of the kind that AT&T, CenturyLink, and Verizon are seeking may or may not be consistent with their own actual practices, it would clearly be inconsistent with the practices of other service providers, and it could seriously disrupt their business and billing practices. Nothing in 47 U.S.C. § 615a-1 suggests that Congress intended such a result.

The absurd impact of BellSouth's faulty "similarly-situated" comparisons on 911 fees can also be illustrated by the following example. Assume that a hypothetical bank in Alabama has three branches, one in the Birmingham 911 District, one in the Calhoun County 911 District, and one in the Autauga County 911 District. Assume further that with IVoIP service, this bank would have the shared simultaneous call capacity of 100 calls across the three branches. Under BellSouth's supposedly "non-discriminatory" reading of 47 U.S.C. § 615a-1, each of these locations should be assessed 911 fees on the basis of what a similarly-situated traditional telephone service customer would pay. To have simultaneous call capacity of 100 calls using traditional telephone service, each of the three branches would have to purchase 5 PRIs (each of which would support 23 channelized access lines). Therefore, under BellSouth's approach, each bank location would be responsible for 100 911 fees—three hundred total fees across all three branches. This example demonstrates that simultaneous call capacity—the method for assessing 911 fees endorsed by BellSouth and other telecom commenters—is not a rational or feasible basis for assessing 911 fees on VoIP. Nor does it accomplish BellSouth's supposed "non-discriminatory" outcome. The impracticality of BellSouth's suggested focus on simultaneous call capacity would be even clearer if this hypothetical bank had IVoIP service from Vonage, Twilio, Telzio, or Mitel that had no simultaneous call limit.

In sum, the telecom commenters want the Commission to find that 47 U.S.C. § 615a-1 requires preemption of state laws that fail to apply the older contractual and technological limitations of traditional telephone services to the much more robust and ever-improving range of IVoIP services. As reiterated more fully below, even in close cases, preemption is highly disfavored in American law, particularly where, as here, traditional state powers are at stake. But this case is nowhere near to being a close one, as the proposed contractual-simultaneous-call-capacity standard is flawed in multiple ways:

- It rests on the demonstrably false premise that traditional and IVoIP services are similarly situated.

- It is based on factual assertions that have not been tested by discovery.

[28] Ex. H, at p. 3 (emphasis added), *available at* https://www.mitel.com/products/business-phone-systems/on-site/other/mitel-sip-overview.

- It may or may not be consistent with AT&T's, CenturyLink's, and Verizon's own contracts.
- It may or may not reflect their own actual practices in handling 911 calls.
- It is inconsistent with the practices of many other telephone service providers and would have significant adverse effects on them.
- It would throw 911 funding into chaos.

Instead of issuing BellSouth's requested declaration, the Commission should find that 47 U.S.C. § 615a-1 simply requires states and local governments to assess the same 911 fee rate for VoIP service and traditional telephone service and does not regulate the total amount of 911 fees that may be charged to a particular customer. Such a declaration would be consistent with the Commission's prior pronouncement in its *E911 IP-Enabled Order*: "Because 911 contribution obligations are typically assessed on a per-line basis, states may need to explore other means of collecting an appropriate amount from competitive LECs on behalf of their interconnected VoIP partners, such as a per-subscriber basis."[29]

States like Alabama have taken the differences between VoIP and traditional service into account when drafting their 911 fee statutes. That was and remains their right to do.

IV. Several States Have Rules that Assess 911 Fees on a Different Basis For IVoIP and Traditional Services.

In their *ex parte* submission, AT&T *et al*. asserted that "no state has adopted a 911 statute that expressly adopts a discriminatory rule." As an initial matter, the term "discriminatory" is nothing more than a pejorative buzzword. IVoIP service and traditional telecommunications service, as explained above, have significant differences that require different treatment. Different treatment based on rational differences in technology is not the type of "discrimination" that the statute prohibits. The idea that all services including those that are functionally distinct must necessarily be treated in an identical manner is a concept manufactured by BellSouth in order to avoid liability for serial under-billing of critical 911 fees.

Setting aside BellSouth's attempt to manufacture a negative connotation, several states have 911 funding statutes that assess 911 fees on IVoIP services and traditional services on a different basis. Specifically, each of the following states assess 911 fees on traditional service by access line and on VoIP by telephone number:

- **Alabama**: prior to 2013, Alabama assessed 911 fees on traditional service by exchange access line (interpreted as voice channel) and VoIP by 10-digit access

[29] *E-911 IP-Enabled Order*, at ¶ 52 n.163.

number. *See* Ala. Code § 11-98-5; *Madison Cty. Commc'ns Dist. v. BellSouth Telecomms., Inc.*, 2009 WL 9087783, at *4-8 (N.D. Ala. Mar. 31, 2009).

- **Florida**: Florida assesses five 911 fees per PRI, up to a maximum of 25 911 fees per account. For IVoIP, Florida assesses one 911 fee per service identifier up to 25 911 fees. Service identifier is defined as "the service number, access line, or other unique identifier assigned to a subscriber." Fla. Stat. § 365.172.
- **Idaho**: Idaho imposes a 911 fee on traditional service by access line and VoIP service by "interconnected VoIP service line," which is defined as "interconnected VoIP service that offers an active telephone number, or successor dialing protocol assigned by a VoIP provider to a VoIP service customer number that has an outbound calling capability of directly accessing a public safety answering point." Idaho Code Ann. § 31-4802.
- **South Carolina**: South Carolina imposes 911 fees on traditional service by "exchange access facility" and on VoIP by "VoIP service line," which is defined as "a VoIP service that offers an active telephone number or successor dialing protocol assigned by a VoIP service provider to a customer that has outbound calling capability." S.C. Code Ann. §§ 23-47-10; 23-47-40; 23-47-67.

BellSouth may disagree with the Districts' interpretation of these statutes, but the statutory language referencing access lines and telephone numbers is undeniable.

Furthermore, contrary to assertions by BellSouth and other telecom commenters, statutory schemes like these will not lead to huge increases in the total amount due by VoIP subscribers for 911 fees. Forty-five states—including Alabama, Florida, Idaho, and South Carolina—have statutes mandating that funds from 911 fees be used solely for 911 purposes.[30] Further, some statutes—including the Alabama's ETSA—require states or local districts to reduce the 911 fee if 911 needs are overfunded. Therefore, if the Commission concludes that preemption is not warranted—which is actually just the status quo—and this leads to more 911 fees being assessed on VoIP service, the likely result will be that states and local districts will lower the rate of the 911 fees in order to account for the broader base of assessed service.

V. BellSouth's Own Filings Belie Its Requested Declaration.

Fundamentally, BellSouth's preemption argument fails the appropriate legal standard. The Supreme Court has plainly held that Congress's intent to preempt traditional state powers must be "unmistakably clear in the language of the statute."[31] This case involves two quintessential areas of traditional state powers—the power to impose taxes and fees and the power to guarantee public safety. Thus, under the *Gregory* standard, Congress's intent to preempt state laws such as ETSA must be "clear and manifest" in the language of 47 U.S.C. § 615a-1.[32] This heightened standard is

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[30] Federal Communications Commission, Tenth Annual Report to Congress On State Collection and Distribution of 911 and Enhanced 911 Fees and Charges (2018), 22–23 *available at* https://www.fcc.gov/files/10thannual911feereporttocongresspdf. [*hereinafter* Commission's 10th Annual Report]; Idaho Code Ann. § 31-4804.

[31] *Gregory v. Ashcroft*, 501 U.S. 452, 460 (1991).

[32] *Gregory*, 501 U.S. at 460.

buttressed by the traditional presumption against preemption—which applies both to whether Congress intended any preemption at all and the scope of the intended preemption.[33] Because preemption of a traditional state power requires unmistakably clear intent and must overcome a presumption against preemption, an ambiguous federal statute is insufficient to preempt. *See Gregory*, 501 U.S. at 470 ("In the face of such ambiguity, we will not attribute to Congress an intent to intrude on state governmental functions . . . .").

BellSouth's basic preemption argument plainly fails to meet this standard because it is based on a finding of ambiguity. Specifically, BellSouth based its request for a primary jurisdiction referral on the notion that 47 U.S.C. § 615a-1 is ambiguous:

> This preemption question is also within the FCC's special expertise. The FCC has authority to decide which VoIP services are subject to 911 obligations, *see* 47 U.S.C. § 615a-1(a), and the expertise necessary to *resolve any ambiguities in the phrases "class of subscribers" and "fee or charge may not exceed" in§ 615a-1(f)(1). Indeed, the FCC has long had regulatory authority* over the "charges . . . in connection with" "classes of communications" service. 47 U.S.C. § 201(b). The FCC's interpretation of § 615a-1(f)(1) will also be informed by the longstanding federal policy of promoting the broadband Internet networks that "make services such as interconnected VoIP service accessible to more Americans." *Connect America Fund Order* ¶¶ 67-69. Finally, *the FCC's interpretation of those ambiguous terms will receive Chevron deference*. *See*, *e.g.*, *Palm Beach Golf Center-Boca, Inc. v. John G. Sarris, D.D.S., P.A.*, 781 F.3d 1245, 1256-57 (11th Cir. 2015) (deferring to FCC statutory interpretation).[34]

BellSouth's basic premise belies its requested finding. Preemption of a traditional state power cannot be based on an ambiguous federal statute. BellSouth asserts that the specific terms in 47 U.S.C. § 615a-1 that it relies on for preemption are ambiguous. Therefore, 47 U.S.C. § 615a-1 cannot preempt traditional state powers because, as BellSouth admits, the intent to preempt is not "unmistakably clear." In other words, BellSouth cannot win based on the premise it advanced in the District Court and is advancing at the Commission. The law requires a finding of no preemption.

Moreover, BellSouth has asserted at various times that the Commission's interpretation of 47 U.S.C. § 615a-1 would be due *Chevron* deference. However, a court only grants *Chevron* deference "[i]f a statute is ambiguous."[35] Therefore, the essential predicate for *Chevron* deference—ambiguity—dooms preemption under the well-established requirement that Congress's intent to preempt a traditional state power must be evidenced by a clear statement.[36]

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[33] *See Medtronic, Inc. v. Lohr*, 518 U.S. 470, 485–86 (1996).
[34] *Autauga Cty. Emergency Mgmt. Commc'n Dist. v. BellSouth Telecomms., LLC*, No. 2:15-cv-00765-SGC (N.D. Ala.). Dkt. 39 at 20–21 (emphasis added).
[35] *National Cable & Telecommunications Ass'n v. Brand X Internet Services*, 545 U.S. 967, 980 (2005).
[36] *See Tennessee v. Federal Communications Commission*, 832 F.3d 597, 615 (6th Cir. 2016) (White, J., concurring in part and dissenting in part) ("But that very silence or ambiguity with respect to whether

VI. BellSouth's Requested Declaration Will Not Reduce Any Compliance Burden.

A finding of preemption will not reduce any regulatory or compliance burden on telephone providers. The Commission has, at times, used its power to preempt state laws to create a more streamlined and centralized regulatory scheme. However, BellSouth's requested preemption declaration will not eradicate state 911 fees. Telephone providers will still have to comply with state 911 fee schemes regardless of any finding by the Commission. In other words, this proceeding is not a tool to streamline compliance or cut red tape.[37]

In fact, BellSouth's proposed interpretation of 47 U.S.C. § 615a-1 would create new—possibly unanswerable—compliance questions. How is VoIP service similarly situated to traditional service in all scenarios? How does a telephone provider account for a burstable number of simultaneous calls? How does a telephone provider bill 911 fees to a customer that shares a call capacity between multiple locations—particularly when a state assesses 911 fees by access line for traditional service? How does a telephone company bill 911 fees in a state with a law that would be preempted by 47 U.S.C. § 615-1? Will states have to issue interim guidance? Will states have to rewrite 911 funding statutes? All of these questions could create significant compliance issues for telephone companies, in addition to confusion for state and local 911 entities and state legislatures.

In short, BellSouth's requested preemption declaration will not streamline compliance with 911 funding statutes, and it will not cut any red tape. Rather, BellSouth's requested declaration has the potential to create more compliance issues and to create massive confusion for state and local 911 entities.

Congress intended to preempt the states' regulation of their subdivisions triggers application of the clear statement rule; thus, application of *Chevron* deference to the FCC's determination that it has the authority to preempt in this situation—as distinguished from the decision whether to exercise the authority to preempt—would turn the clear statement rule on its head.").

[37] The Districts do not contend that 911 fees are "red tape." 911 funding is essential to public safety, and the services funded by 911 fees save lives on a daily basis.

Yours truly,

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Exhibit A

verizon✓

VOICE OVER IP SERVICE

1. GENERAL

1.1 **Service Definition.** Voice over IP (VoIP) Service enables Customer to make telephone calls via the Internet. Verizon offers two types with Optimized and Non-Optimized Service: IP Integrated Access (for sites with key or PBX systems) and IP Trunking (for IP PBX equipment); and a third type with Non-Optimized VoIP Service: Hosted IP Centrex, where all the features of a PBX or key system reside on Verizon's VoIP network.

- **Platforms**. Except where explicitly stated otherwise, these terms apply to Optimized VoIP + Service (denoted with a "+" and sometimes referred to as Rapid Delivery) and non-Optimized VoIP Service.

1.2 Standard Service Features

1.2.1 **Calling Capacity.** With VoIP Service, Verizon provides Customer the ability to select its simultaneous calling capacity.

1.2.2 **Burstable Enterprise Shared Trunks (BEST).** Customer's VoIP sites that are provisioned with BEST will be able to share the total simultaneous calling capacity purchased by Customer across its enterprise on a regional basis. Thus, simultaneous call units within a region contribute to the total available concurrent call capacity only within that region. Concurrent call pools cannot be regionally shared between the U.S./Canada, Europe, and Asia-Pac regions. BEST applies to enterprises in which all locations are on a metered or tiered pricing model. Simultaneous calling capacity can be shared between locations receiving both Local and LD VoIP service, and between locations receiving only LD service, but not across those two kinds of locations.

1.2.3 **Local/National Calling Services**

1.2.3.1 **Outbound Public Service Telephone Network (PSTN or Local) Calls.** Verizon enables Customer to place calls to most PSTN destinations, including but not limited to, local, national, international, fixed-to-mobile, Directory Assistance and non-geographic destinations. For Europe, a list of destinations not currently supported by VoIP Service is available upon Customer's request.

1.2.3.2 **Number Portability.** Verizon enables Customer to port its telephone numbers (i.e., retain them) using Local Number Portability (LNP) at the same time VoIP Service is made available for use, or delay LNP for up to 10 days afterwards.

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VoIP minutes.

If simultaneous calling units are provisioned at the location level (level available with Non-Optimized VoIP Service and Optimized VoIP Service), a minimum of one unit must be purchased for each location and allotted minutes cannot be shared between locations, nor can they be rolled over from month to month.

If the simultaneous calling capacity is provisioned at the enterprise level (level available with Optimized VoIP Service), minutes can be shared between Customer locations (with like Services, e.g., Local and LD to Local and LD), but they cannot be rolled over from month to month. Tiered simultaneous calling units cannot be provisioned at the enterprise level in the Europe and Asia-Pac regions.

Calls to international locations can also be made but are billed at metered rates.

5.1.2.2 **Metered Pricing – Simultaneous Calling Capacity Charge.** Customer will pay an MRC per simultaneous calling unit multiplied by the number of simultaneous call units Customer selects. Each such simultaneous calling unit includes:

- unlimited intra-enterprise VoIP calls (VoIP origination and termination within Customer's enterprise), and
- for U.S./Canada VoIP locations, local calling if Local service is offered in the affected region and purchased by Customer.

Inter-enterprise VoIP calls (termination is outside Customer's enterprise), including LD or national calls, as applicable, are billed a per-minute charge. Calls to international locations can also be made but are billed at metered rates.

Simultaneous calling units can be provisioned for metered pricing at both the location and enterprise levels for Optimized Service and at the location level for Non-Optimized Service. If simultaneous calling units are provisioned at the location level, a minimum of one unit must be purchased for each hub and remote location.

5.1.2.3 **Virtual Communication Express over VoIP Pricing - Simultaneous Calling Capacity Charge.** Customer will pay an MRC per simultaneous calling unit multiplied by the number of simultaneous call units Customer selects when Customer implements Virtual Communication Express over VoIP at sites in the U.S. Each such simultaneous calling unit includes:

- unlimited intra-enterprise VoIP calls (VoIP origination and termination within Customer's enterprise); and
- Unlimited U.S. domestic LD minutes and unlimited Local calling if Local Service is offered in the affected region and purchased by Customer.

Virtual Communications Express over VoIP may only be installed in sites in the U.S. Unlimited concurrent calls is only available when the U.S. site (i) uses location level concurrent calls; and (ii) implements Virtual Communication Express with VoIP.

5.1.2.4 **BEST+.** BEST+ is an optional billable feature available if Customer (i) purchases Optimized VoIP Service via a "right to buy" arrangement, and (ii) purchases a minimum of 200 simultaneous calling units at the enterprise level. With BEST+, Customer can exceed (or "burst") its simultaneous calling capacity if, for example, it experiences an unplanned burst of inbound/outbound voice calls. To enable BEST+, Customer will be charged an MRC based on its simultaneous calling capacity purchased at the enterprise level and its selected tier of burstable simultaneous calling units (see table below). Customer also will be charged an NRC for the maximum number of bursted simultaneous calling units attained during the affected billing period.

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Exhibit B

AT&T IP Flexible Reach
Enterprise Administrator Guide

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About Trunk Burst

Bursting and sharing is functionality that allows for calls that go beyond the maximum number allowed on the trunk to borrow call capacity from another group trunk group within your enterprise.

The maximum group trunk group burst can't exceed 20% of a group trunk group's maximum concurrent call limit. Bursting and sharing is initially set to false and is managed by AT&T. If your Enterprise wants to utilize or make changes to this functionality, contact your AT&T account representative.

The Trunk Burst setting on a group trunk group indicates whether your Enterprise is using this functionality or not. The setting is either true or false.

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Exhibit C

CHAPTER 15

Emergency Services

Revised: June 14, 2016

Emergency services are of great importance in the proper deployment of a communications system. This chapter presents a summary of the following major design considerations essential to planning for emergency calls:

This chapter presents some information specific to the 911 emergency networks as deployed in Canada and the United States. Many of the concepts discussed here are adaptable to other locales. Please consult with your local telephony network provider for appropriate implementation of emergency call functionality.

In the United States, some states have already enacted legislation covering the 911 functionality required for users in a multi-line telephone system (MLTS). The National Emergency Number Association (NENA) has also produced the *NENA Technical Requirements Document on Model Legislation E9-1-1 for Multi-Line Telephone Systems*, available online at

http://www.nena.org/

This chapter assumes that you are familiar with the generic 911 functionality available to residential PSTN users in North America.

Note The topics discussed in this chapter apply to Cisco Emergency Responder only when it is used in conjunction with Cisco Unified Communications Manager (Unified CM). Cisco TelePresence Video Communication Server (VCS) currently does not support emergency services.

What's New in This Chapter

Table 15-1 lists the topics that are new in this chapter or that have changed significantly from previous releases of this document.

Table 15-1 New or Changed Information Since the Previous Release of This Document

New or Revised Topic	Described in	Revision Date
Service Provider ALI (SP-ALI)	Service Provider ALI, page 15-3	June 14, 2016
SIP trunk support	Dynamic ANI (Trunk Connection), page 15-7	June 14, 2016
Access point tracking	Device Location Discovery Methods in Cisco Emergency Responder, page 15-10	June 14, 2016
Location awareness for wireless clients	Cisco Emergency Responder and Location Awareness for Wireless Clients, page 15-19	June 14, 2016
Other minor updates	Various sections of this chapter	June 14, 2016
Cisco Unified Communications Manager Native Emergency Call Routing	Emergency Call Routing Using Unified CM Native Emergency Call Routing, page 15-27	June 15, 2015

911 Emergency Services Architecture

This section highlights some of the functionality requirements for emergency calls in multi-line telephone systems (MLTS). In the context of this section, emergency calls are 911 calls serviced by the North American public switched telephone network (PSTN).

Any emergency services architecture usually consists of the following elements:

- A distressed caller should be able to dial the emergency services from a fixed line, a mobile phone, a public phone, or any device capable of making the voice call.
- An emergency services call handler must be available to respond to the emergency request and dispatch the needed services such as police, fire, and medical.
- In order to provide help, the call handler should be able to identify the location of the distressed caller as precisely as possible.
- An emergency services network is needed to route the call to the nearest emergency services call handler with jurisdiction for the location of the caller.

The following sections explain some of the important architectural components of 911 emergency services architecture.

Public Safety Answering Point (PSAP)

The public safety answering point (PSAP) is the party responsible for answering the 911 call and arranging the appropriate emergency response, such as sending police, fire, or ambulance teams. The physical location of the phone making the 911 call is the primary factor in determining the appropriate PSAP for answering that call. Generally, each building is serviced by one local PSAP.

To determine the responsible PSAP for a given location, contact a local public safety information service such as the local fire marshal or police department. Also, the phone directory of the local exchange carrier usually lists the agency responsible for servicing 911 calls in a given area.

Typical Situation

- For a given street address, there is only one designated PSAP.
- For a given street address, all 911 calls are routed to the same PSAP.

Exceptional Situation

- The physical size of the campus puts some of the buildings in different PSAP jurisdictions.
- Some of the 911 calls need to be routed to an on-net location (campus security, building security).

Selective Router

The selective router is a node in the emergency services network that determines the appropriate PSAP for call delivery, based on caller's geographic area and the automatic number identification (ANI). The Local Exchange Carrier (LEC) usually operates the selective router. Hence, it is imperative to ensure that the enterprise IP communications network is designed in such a way that the caller is routed to the appropriate selective router based on its location.

Automatic Location Identifier Database

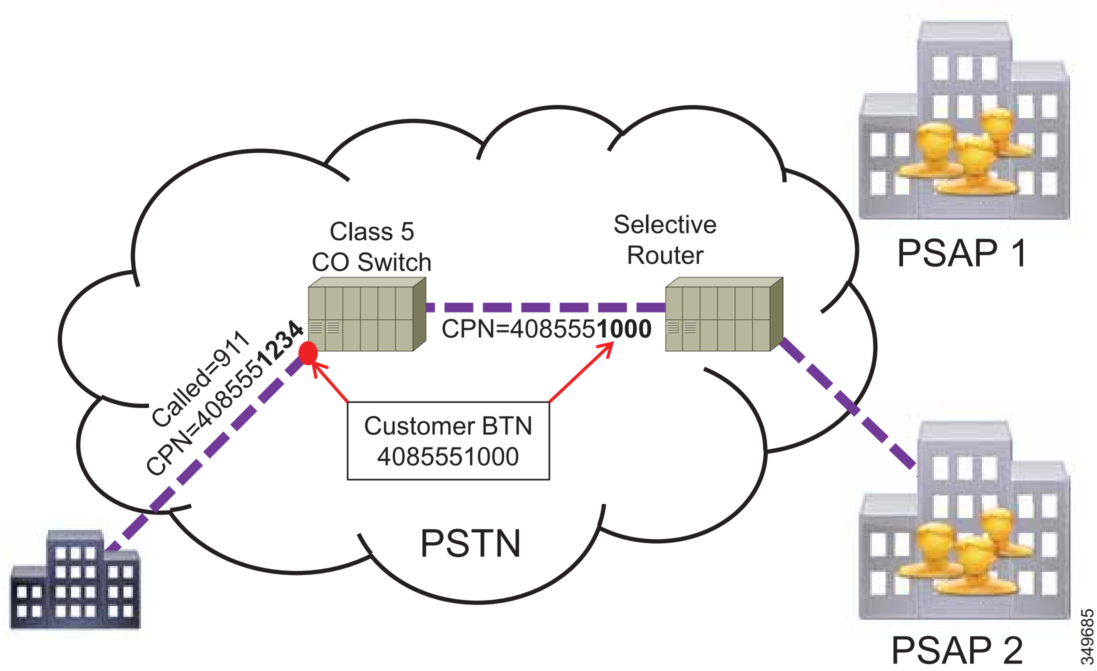
Location information of the caller is an important part of the 911 services infrastructure. The Automatic Location Identifier (ALI) database maintains the location information for the particular geographical location served by the LEC. For every 911 call, the PSAP searches the ALI database to retrieve the caller's location based on the ANI of the calling number. The addresses are stored in the Master Street Address Guide (MSAG) format in the ALI database. The ALI database is maintained on behalf of the local emergency services administration by a contracted third party, generally the incumbent Local Exchange Carrier (LEC).

Service Provider ALI

Service Provider ALI (SP-ALI) refers to a configuration in which the service provider is responsible for defining and maintaining the ALI information for all emergency calls over the connection. SP-ALI service uses the physical interconnection at the LEC to determine the source location of the call. For residential customers, the ALI information is associated with the address of the subscriber and the directory number of that resident. Because the ALI information is determined by the service provider based upon the physical interconnection in the LEC, the subscriber does not have the ability to change or set the ALI information.

The setting of the ALI information based on the physical point of interconnection of the line or trunk applies to PRI trunk connections also. By default, an MLTS operator that uses PRI trunks for PSTN access will have SP-ALI service. The LEC defines the calling party number (CPN) and ALI address for emergency calls. Typically, the calling party number used for emergency calls is the customer's bill-to number (BTN) or the MLTS operator's main number. The physical address associated with the emergency calling number is the address of the demark of the PRI at the customer's facility. If a PRI trunk is set for SP-ALI service, all calls to 911 have the calling party number replaced by the LEC to match the ALI record for the customer. (See Figure 15-1.)

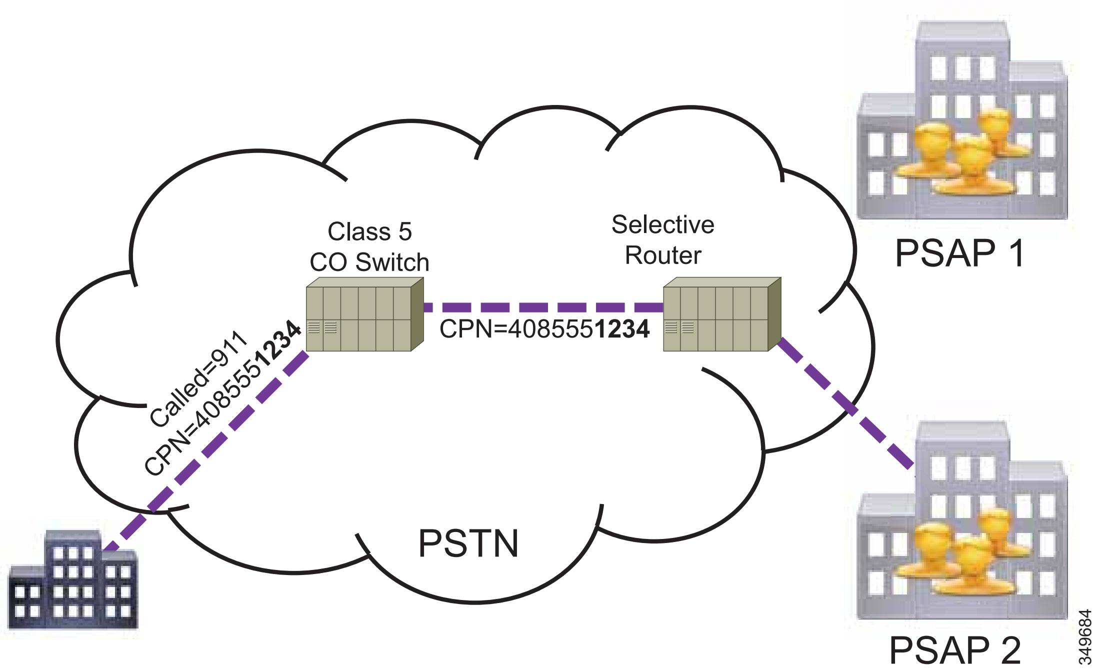
Figure 15-1 Service Provider ALI



Private Switch ALI

Private Switch ALI (PS-ALI) is an enhancement to 911 emergency response systems that enables MLTS operators to provide more specific address and location information for each endpoint. The service allows a customer-generated address table to be loaded into the ALI database so that each station of an MLTS system can be uniquely identified if a call is placed to 911 from that telephone number. The station-specific or location-specific automatic number identification (ANI) generated by the communications system can be passed directly to the E911 system to pinpoint the precise location of the caller. (See Figure 15-2.) The PSAP operator can then direct emergency response personnel to the correct address, building, floor, room, or even cubicle, thereby streamlining operations and increasing accuracy.

Figure 15-2 Private Switch ALI



911 Network Service Provider

After identifying the responsible PSAPs, you must also identify the 911 network service providers to which each PSAP is connected. It is commonly assumed that PSAPs receive 911 phone calls from the PSTN, but that is not the case. Instead, 911 calls are carried over dedicated, regionally significant networks, and each PSAP is connected to one or more such regional networks. In the majority of cases, the incumbent Local Exchange Carrier (LEC) is the 911 network service provider for a PSAP. Some exceptions include military installations, university campuses, federal or state parks, or other locations where the public safety responsibility falls outside the jurisdiction of the local authorities and/or where a private network is operated by an entity other than a public local exchange carrier.

If you are in doubt about the 911 network service provider for a given PSAP, contact the PSAP directly to verify the information.

Typical Situation

- For a given street address, the 911 network service provider is the incumbent Local Exchange Carrier (LEC). For a location served by Phone Company X, the corresponding PSAP is also served by Phone Company X.
- All 911 calls are routed directly to an off-net location, or all 911 calls are routed directly to an on-net location.

Exceptional Situation

- The local exchange carrier (LEC) through which the MLTS interfaces to the PSTN is *not* the same LEC that serves as 911 network service provider to the PSAP. (For example, the communications system is served by Phone Company X, but the PSAP is connected to Phone Company Y.) This situation might require either a special arrangement between the LECs or special, dedicated trunks between the phone system and the PSAP's 911 network service provider.
- Some LECs may not accept 911 calls on their networks. If this is the case, the only two options are to change LECs or to establish trunks (dedicated to 911 call routing) connected to a LEC that can route 911 calls to the appropriate PSAPs.
- Some (or all) of the 911 calls have to be routed to an on-net location such as campus security or building security. This situation can easily be accommodated during the design and implementation phases, but only if the destination of 911 calls for each phone has been properly planned and documented.

Interface Points into the Appropriate 911 Networks

For larger communications systems, 911 connectivity might require many interface points. Typically, more than one E911 selective router is used within a LEC's territory, and these routers usually are *not* interconnected.

For example, an enterprise with a large campus could have the following situation:

- Building A located in San Francisco
- Building B located in San Jose
- San Francisco Police Department and San Jose Police Department are the appropriate PSAPs
- San Francisco Police Department and San Jose Police Department are served by the same 911 network service provider
- However, San Francisco Police Department and San Jose Police Department are served by different E911 selective routers operated by that same 911 network service provider!

This type of situation would require two separate interface points, one per E911 selective router. The information pertaining to the E911 selective router territories is generally kept by the incumbent LEC, and the local account representative for that LEC should be able to provide an enterprise customer with the pertinent information. Many LECs also provide the services of 911 subject matter experts who can consult with their own account representatives on the proper mapping of 911 access services.

Typical Situation

- For single-site deployments or campus deployments, there is usually only one PSAP for 911 calls.
- If access to only one PSAP is required, then only one interface point is required. Even if access to more than one PSAP is required, they might be reachable from the same E911 selective router, through the same centralized interface. If the enterprise's branch sites are linked via a WAN (centralized call processing), it is desirable to give each location its own local (that is, located inside each branch office) access to 911 to prevent 911 isolation during WAN failure conditions where Survivable Remote Site Telephony (SRST) operation is activated.

Exceptional Situation

- The physical size of the campus puts some of the buildings in different PSAP jurisdictions, *and*
- Some of the 911 calls have to be routed to different E911 selective routers, through different interface points.

Note Some of the information required to establish the geographical territories of PSAPs and E911 selective routers is available online or from various competitive local exchange carrier (CLEC) information web sites. (For example, https://clec.att.com/clec/hb/shell.cfm?section=782 provides some valuable data about the territory covered by AT&T in California and Nevada.) However, Cisco strongly recommends that you obtain proper confirmation of the appropriate interface points from the LEC prior to the design and implementation phases of 911 call routing.

Interface Type

In addition to providing voice communications, the interfaces used to present 911 calls to the network must also provide identification data about the calling party.

Automatic Number Identification (ANI) refers to the North American Numbering Plan number of the calling party, which is used by networks to route a 911 call to the proper destination. This number is also used by the PSAP to look up the Automatic Location Identification (ALI) associated with a call.

911 calls are source-routed, which means that they are routed according to the calling number. Even though different locations are all dialing the same number (911), they will reach different PSAPs based on their location of origin, which is represented by the ANI (calling number).

You can implement 911 call functionality with either of the following interface types:

- Dynamic ANI assignment
- Static ANI assignment

While dynamic ANI assignment scales better (because it supports multiple ANIs) and lends itself to all but the smallest of applications, static ANI assignment can be used in a wider variety of environments, from the smallest to the largest systems.

Dynamic ANI (Trunk Connection)

The dynamic aspect of ANI refers to the fact that a communications system has many endpoints sharing access to the 911 network across the same interface, and the ANI transmitted to the network might need to be different for each call.

There are three main types of dynamic ANI interfaces:

- Integrated Services Digital Network Primary Rate Interface (ISDN-PRI, or simply PRI)
- Session Initiation Protocol (SIP) trunk
- Centralized Automatic Message Accounting (CAMA).

PRI

This type of interface usually connects a communications system to a PSTN Class 5 switch. The calling party number (CPN) is used at call setup time to identify the E.164 number of the calling party.

Most LECs treat the CPN differently when a call is made to 911. Depending upon the functionality available in the Class 5 switch and/or upon LEC or government policy, the CPN may not be used as the ANI for 911 call routing. Instead, the network may be programmed to use the listed directory number (LDN) or the bill-to number (BTN) for ANI purposes.

If the CPN is not used for ANI, then 911 calls coming from a PRI interface all look the same to the 911 network because they all have the same ANI, and they are all routed to the same destination (which might not be the appropriate one). The replacement of the CPN by the LEC is typically called Service Provider ALI (SP-ALI), because the service provider specifies the CPN for ALI lookup.

Some LECs offer a feature to provide CPN transparency through a PRI interface for 911 calls. With this feature, the CPN presented to the Class 5 switch at call setup is used as ANI to route the call. The feature name for this functionality varies, depending on the LEC. (For example, SBC calls it Inform 911 in California.)

Note When SP-ALI service is used, the CPN *must* be a routable North American Numbering Plan number, which means that the CPN must be entered in the routing database of the associated E911 selective router.

Note For Direct Inward Dial (DID) phones, the DID number could be used as the ANI for 911 purposes, but only if it is properly associated with an Emergency Service Number in the 911 service provider's network. For non-DID phones, use another number. (See Emergency Location Identification Number Mapping, page 15-14, for more information.)

Many Class 5 switches are connected to E911 selective routers through trunks that do not support more than one area code. In such cases, if PRI is used to carry 911 calls, then the only 911 calls that will be routed properly are those whose CPN (or ANI) have the same Numbering Plan Area (NPA) as the Class 5 switch.

Example

An MLTS is connected to a Class 5 switch in area code 514 (NPA = 514). If the MLTS were to send a 911 call on the PRI trunk, with a CPN of **450**.555.1212, the Class 5 switch would send the call to the E911 selective router with an ANI of **514**.555.1212 (instead of the correct **450**.555.1212), yielding inappropriate routing and ALI lookup.

To use PRI properly as a 911 interface, the system planner must ensure that the CPN will be used for ANI and must properly identify the range of numbers (in the format NPA NXX TNTN) acceptable on the link. For example, if a PRI link is defined to accept ANI numbers within the range 514 XXX XXXX, then only calls that have a Calling Party Number with NPA = 514 will be routed appropriately.

SIP Trunk

SIP trunking is an IP-only interface that connects a communications system to a service provider, typically through a Session Border Controller (SBC). SIP trunks allow for the same dynamic calling party number delivery to the carrier as PRI trunks; but unlike PRI trunks, SIP trunks do not have a physical limit on the number of calls that can be established concurrently.

When emergency services are called over a SIP trunk, delivery of the call to the correct selective router must be verified with the provider. Unlike PRI circuits that terminate at the local LEC, SIP trunks might not have a physical connection with the local LEC and as a result will not automatically route 911 calls to the selective router in the municipality of the calling party. Additionally, each SIP trunk provider might have different E911 routing capability; for example, one service provider may be able to deliver calls to selective routers across the US based upon the calling party number (even outside the local area), while another service provider may allow E911 calls into only one customer-specified selective router. A Cisco Unified CM administrator should always confirm the 911 call delivery capabilities with the carrier, especially when a SIP trunk is providing centralized call routing.

SIP service providers are required to route 911 calls to the appropriate rate center or PSAP for any DID number that they service over a SIP trunk. For example, assume that a deployment has a SIP trunk that physically terminates in a data center in Dallas Texas that services DIDs for a San Francisco office with the range of 415-555-1xxx and for a New York office with the range of 212-448-2xxx. If a call to 911 is place from 415-555-1800, then the SIP provider must route the call to the San Francisco selective router for PSAP delivery. If a user at extension 212-448-2840 in the New York City office dials 911, the call can be routed on the same SIP trunk to the appropriate selective router in the New York City area to reach the PSAP appropriate for the caller.

CAMA

Centralized Automatic Message Accounting (CAMA) trunks also allow the MLTS to send calls to the 911 network, with the following differences from the PRI approach:

- CAMA trunks are connected directly into the E911 selective router. Extra mileage charges may apply to cover the distance between the E911 selective router and the MLTS gateway point.
- CAMA trunks support 911 calls only. The capital and operational expenses associated with the installation and operation of CAMA trunks support 911 traffic only.
- CAMA trunks for the MLTS market may be limited to a fixed area code, and the area code is typically implied (that is, not explicitly sent) in the link protocol. The connection assumes that all calls share the same deterministic area code, therefore only 7 or 8 digits are sent as ANI.

Static ANI (Line Connection)

Static ANI provides a line (rather than a trunk) connection to the PSTN, and the ANI of the line is associated with all 911 calls made on that line, regardless to the CPN of the calling phone. Static ANI is based on the physical interconnection point in the LEC. Because the Static ANI is defined by the carrier on the interconnection point in the LEC, Static ANI emergency call routing is also referred to as Service Provider ALI (SP-ALI). A plain old telephone service (POTS) line is the most common type of connection used for this purpose.

POTS lines are one of the simplest and most widely supported PSTN interfaces. A POTS line usually comes fully configured to accept 911 calls. In addition, the existing E911 infrastructure supports 911 calls from POTS lines very well.

The POTS approach has the following attributes:

- The operational costs associated with a POTS line are low.
- The POTS line can even serve as a backup line in case of power failure.
- The POTS line number can be used as the callback number entered into the ALI database.
- POTS lines represent the lowest cost 911 support for locations where user density does not justify local PRI or CAMA access into the PSTN.
- POTS lines are ubiquitous in PSTN installations.

All outgoing 911 calls through this type of interface are treated the same by the E911 network, and any tools that enable ANI manipulation presented to the E911 network (such as translations or transformations) are irrelevant because the ANI can be only the POTS line’s number.

Cisco Emergency Responder

Ease of administration for moves, adds, and changes is one of the key advantages of IP communications technology. To provide for moves, adds, and changes that automatically update 911 information without user intervention, Cisco has developed a product called the Cisco Emergency Responder (Emergency Responder).

Cisco Emergency Responder provides the following primary functionality:

- Dynamic association of a phone to an Emergency Response Location (ERL), based on the detected physical location of the phone.
- Dynamic association of the Emergency Location Identification Number (ELIN) to the calling phone, for callback purposes. In contrast to the general emergency services scenarios outlined in preceding sections, Cisco Emergency Responder enables the callback to ring the exact phone that initiated the 911 call.
- On-site notification to designated parties (by pager, web page, email, or phone call) to inform them that there is an emergency call in progress. Email, pager, and web page notifications include the calling party name and number, the ERL, and the date and time details associated with the call. Phone notification provides the information about the calling number from which the emergency call was placed.

For more information on ERLs and ELINs, see Emergency Response Location Mapping, page 15-13, and Emergency Location Identification Number Mapping, page 15-14. For more information on Cisco Emergency Responder, see Cisco Emergency Responder Design Considerations, page 15-19, and refer to the Cisco Emergency Responder product documentation available online at

http://www.cisco.com/en/US/products/sw/voicesw/ps842/tsd_products_support_series_home.html

Device Location Discovery Methods in Cisco Emergency Responder

Cisco Emergency Responder uses multiple methods to determine the physical location of a device. Because more specific location discovery results in a shorter time to locate the emergency and administer emergency services, Emergency Responder uses the following methods (listed in priority order) to identify an emergency caller's location:

1. Switch port discovery
2. Access point association
3. IP subnet
4. Static DN assignment
5. Default route

Switch Port Discovery

The primary method for location identification in Cisco Emergency Responder is the detection of an endpoint via Layer 2 discovery at the switch port level. Discovering an endpoint though Layer 2 Cisco Discovery Protocol (CDP) discovery enables Emergency Responder to determine the exact physical location of the calling device based on the physical termination of the network cable to a network jack in a cubicle or office. Although the discovery mechanism of the connected device is reliable, the accuracy of the physical location relies on two main assumptions:

- The wired infrastructure of the enterprise is well established and does not change sporadically, and any wiring closet changes trigger notification to the Emergency Responder administrator indicating what changed.
- The infrastructure is available for Cisco Emergency Responder to browse; that is, Cisco Emergency Responder can establish Simple Network Management Protocol (SNMP) sessions to the underlying network infrastructure and can scan the network ports for the discovery of connected phones.

Once Cisco Emergency Responder discovers the originating port for the call, it associates the call with the pre-established ERL for the location of that port. This process also yields an association with a pre-established ELIN for the location and the selection of the appropriate egress point to the E911 infrastructure, based on the originating ERL.

Access Point Association

Because wireless devices do not have the same discovery capability and tracking characteristics as a wired endpoint, Cisco Emergency Responder tracks wireless clients by using the Location Awareness feature available in Unified CM 11.5 and later releases. The Location Awareness feature allows Emergency Responder to synchronize all deployed access points in Unified CM and to assign the APs to the appropriate ERL. The Location Awareness feature also allows for the updating of mobile device movement between APs.

Emergency Responder is able to track wireless clients across the enterprise through the Location Awareness feature in Unified CM. When a mobile client associates with an AP in the enterprise, the device sends the Basic Service Set Identifier (BSSID) of the AP to Unified CM through call control. Unified CM then updates the database with the new AP association. Periodically, Emergency Responder requests device updates from Unified CM for any device that has updated its AP association since the last request. Emergency Responder receives only the devices that have moved since the last request. In Unified CM 11.5, the request interval is 2 minutes.

IP Subnet

Cisco Emergency Responder also provides the capability to configure ERLs for IP subnets and to assign IP endpoint location by IP address. This capability may be used to locate wireless IP phones, IP softphones, collaboration endpoints that do not support Cisco Discovery Protocol (CDP), and third-party SIP endpoints registered to Cisco Unified CM, which Cisco Emergency Responder cannot locate by connected switch port. It may also be used instead of, or in addition to, connected switch port locations for wired Cisco Collaboration endpoints. If both connected switch port and IP subnet locations are available for a Cisco Collaboration endpoint, Cisco Emergency Responder will prefer the connected switch port location because it is usually more specific than the IP subnet location. Using both connected switch port and IP subnet locations is a best practice because it provides assurance that an appropriate ERL will be assigned, even in case of any delay or error in detecting the connected switch port.

Cisco Emergency Responder allows for the use of two or more ELINs per ERL. The purpose of this enhancement is to cover the specific case of more than one 911 call originating from a given ERL within the same general time period, as illustrated by the following examples.

Example 1

- Phone A and phone B are both located within ERL X, and ERL X is associated with ELIN X.
- Phone A makes a 911 call at 13:00 hours. ELIN X is used to route the call to PSAP X, and PSAP X answers and releases the call. Then, at 13:15 hours, phone B makes a 911 call. ELIN X is again used to route the call to PSAP X.
- PSAP X, after releasing the call from phone B, decides to call back phone A for further details pertaining to phone A's original call. The PSAP dials ELIN X, and gets phone B (instead of the desired phone A).

To work around this situation, Cisco Emergency Responder allows you to define a pool of ELINs for each ERL. This pool provides for the use, in a round-robin fashion, of a distinct ELIN for each successive call. With the definition of two ELINs for ERL X in our example, we now have the situation described in Example 2.

Example 2

- Phone A and phone B are both located within ERL X. ERL X is associated with both ELIN X1 and ELIN X2.
- Phone A makes a 911 call at 13:00 hours. ELIN X1 is used to route the call to PSAP X, and PSAP X answers and releases the call. Then, at 13:15 hours, phone B makes a 911 call, and ELIN X2 is used to route this call to PSAP X.
- PSAP X, after releasing the call from phone B, decides to call back phone A for further details pertaining to phone A's original call. The PSAP dials ELIN X1 and gets phone A.

Of course, if a third 911 call were made but there were only two ELINs for the ERL, the situation would allow for callback functionality to properly reach only the last two callers in the sequence.

High Availability for Emergency Services

It is very important for emergency services to always be available to the user even under the most critical conditions. Therefore, high availability planning must be done carefully when deploying emergency services in an enterprise.

Cisco Emergency Responder supports clustering with a maximum of two servers in active/standby mode. The data is synchronized between the primary and the secondary Cisco Emergency Responder servers. To ensure that calls are routed to the secondary server if the primary server is unavailable, the system administrator must follow certain provisioning guidelines for configuring CTI route points and the directory numbers (DNs) associated to those CTI route points in Cisco Unified CM. For more details on configuration, refer to the *Cisco Emergency Responder Administration Guide*, available at

http://www.cisco.com/en/US/products/sw/voicesw/ps842/prod_maintenance_guides_list.html

If both of the Cisco Emergency Responder servers are unavailable, a local route group (LRG) may be used to route the call to the appropriate PSAP with an appropriate ELIN/ERL (which might be less specific than what Cisco Emergency Responder could have provided). Alternatively, the call may be routed to an internal security office to determine the caller's location. In either case, this provisioning must be done in Cisco Unified CM.

Apart from Cisco Emergency Responder redundancy, Cisco Unified CM redundancy and gateway/trunk redundancy should also be considered to route the 911 emergency calls and to avoid any single point of failure.

Capacity Planning for Cisco Emergency Responder Clustering

In a Cisco Emergency Responder cluster, the quantity of endpoints roaming outside the tracking domain of their home Cisco Emergency Responder group is a scalability factor that must be kept within the limits set forth in the section on *Network Hardware and Software Requirements* in the *Cisco Emergency Responder Administration Guide*, available at:

http://www.cisco.com/en/US/products/sw/voicesw/ps842/prod_maintenance_guides_list.html

For deployments that exceed the Emergency Responder maximum roaming capacity limit (for instance, large campus deployments with multiple Unified CM clusters), phone movement can be tracked by IP subnets. By defining the IP subnets in each of the Cisco Emergency Responder groups and by assigning each ERL with one ELIN per Cisco Emergency Responder group, you can virtually eliminate roaming phones because all phones in the campus will be part of the tracking domain of their respective Cisco Emergency Responder group.

To ensure proper sizing, use the Cisco Unified Communications Sizing Tool (Unified CST). This tool is available only to Cisco partners and employees, with appropriate login required, at http://cucst.cloudapps.cisco.com/landing. If you do not have access to this sizing tool, work with your Cisco account team or partner integrator to size your system appropriately.

Design Considerations for 911 Emergency Services

When planning 911 emergency services for multi-line telephone system (MLTS) deployments, first establish all of the physical locations where phone services are needed. The locations can be classified as follows:

- Single building deployments, where all users are located in the same building
- Single campus deployments, where the users are located in a group of buildings situated in close proximity
- Multisite deployments, where users are distributed over a wide geographical area and linked to the call processing site through WAN connectivity

The locations, or type of deployment, affect the criteria used to design and implement 911 services. The following sections describe the key criteria, along with typical and exceptional situations for each. When analyzing and applying these criteria, consider how they are affected by the phone locations in your network.

Emergency Response Location Mapping

The National Emergency Number Association (NENA) has proposed model legislation to be used by state and federal agencies in enacting the rules that govern 911 in enterprise communications systems. One of the concepts in the NENA proposal is that of the emergency response location (ERL), which is defined as:

> *A location to which a 911 emergency response team may be dispatched. The location should be specific enough to provide a reasonable opportunity for the emergency response team to quickly locate a caller anywhere within it.*

Rather than having to identify each endpoint's location individually, the requirement allows for the grouping of endpoints into a "zone," the ERL. The maximum size of the ERL may vary, depending upon local implementation of the legislation, but we will use 7000 square feet (sq ft) as a basis for discussion in this section. (The concepts discussed here are independent of the maximum ERL size that may be allowed in any given state or region.)

An emergency location identification number (ELIN) is associated with each ERL. The ELIN is a fully qualified E.164 number, used to route the call within the E911 network. The ELIN is sent to the E911 network for any 911 call originating from the associated ERL. This process allows more than one phone to be associated with the same fully qualified E.164 number for 911 purposes, and it can be applied to DID and non-DID phones alike.

Note This document does not attempt to present the actual requirements of any legislation. Rather, the information and examples presented here are for the purposes of discussion only. The system planner is responsible for verifying the applicable local requirements.

For example, assume a building has a work area of 70,000 sq ft and 100 endpoints. In planning for 911 functionality, the building can be divided into 10 zones (ERLs) of 7000 sq ft each, and each endpoint can be associated with the ERL where it is located. When a 911 call is made, the ERL (which could be the same for multiple endpoints) is identified by sending the associated ELIN to the PSAP. If the endpoints were evenly distributed in this example, each group of 10 endpoints would have the same ERL and, therefore, the same ELIN.

The various legislations define a minimum number of endpoints (for example, 49) and a minimum work area (for example, 40,000 sq ft) below which the requirements for MLTS 911 are not applicable. But even if the legislation does not require 911 functionality for a given enterprise, it is always best practice to provision for it.

Emergency Location Identification Number Mapping

In general, you must associate a single fully qualified E.164 number, known as the emergency location identification number (ELIN), with each ERL. (However, if using Cisco Emergency Responder, you can configure more than one ELIN per ERL.) The ELIN is used to route the call across the E911 infrastructure and is used by the PSAP as the index into the ALI database.

ELINs must meet the following requirements:

- The ELIN must be routable across the E911 infrastructure. (See the examples in the section on Interface Type, page 15-7.) If an ELIN is not routable, 911 calls from the associated ERL will, at best, be handled according to the default routing programmed in the E911 selective router.
- Once the ERL-to-ELIN mapping of an enterprise is defined, the corresponding ALI records must be established with the LEC so that the ANI and ALI database records serving the PSAP can be updated accurately.
- The ELIN must be reachable from the PSAP for callback purposes.

The ELIN mapping process can be one of the following, depending on the type of interface to the E911 infrastructure for a given ERL:

- Dynamic ANI interface

  With this type of interface, the calling party number identification passed to the network is controlled by the MLTS. The telephony routing table of the MLTS is responsible for associating the correct ELIN with the call, based on the calling endpoint's ERL. In scenarios where Cisco Emergency Responder is not deployed, the calling party number for calls made to 911 can be modified by Unified CM using transformation masks. For example, all endpoints located in a given ERL can share the same calling search space that lists a partition containing a translation pattern (911) and a calling party transformation mask that would replace the endpoint's CPN with the ELIN for that location. On the other hand, if Cisco Emergency Responder is deployed, calling party number modification should be done on the Emergency Responder system.

- Static ANI interface

  With this type of interface, the calling party number identification passed to the network is controlled by the PSTN. This is the case if the interface is a POTS line. The ELIN is the phone number of the POTS line, and no further manipulation of the phone's calling party identification number is possible.

PSAP Callback

The PSAP might have to reach the caller after completion of the initial conversation or if the caller hangs up before the PSAP operator answers the call. The PSAP's ability to call back relies on the information that it receives with the original incoming call.

The delivery of this information to the PSAP is a two-part process:

1. The Automatic Number Identification (ANI) is first sent to the PSAP. The ANI is the E.164 number used to route the call. In our context, the ANI received at the PSAP is the ELIN that the MLTS sent.
2. The PSAP then uses the ANI to query a database and retrieve the Automatic Location Identification (ALI). The ALI provides the PSAP attendant with information such as:
   - Calling company name
   - Physical address
   - Applicable public safety agency
   - Other optional information, which could include callback information. For example, the phone number of the enterprise's security service could be listed, to aid in the coordination of rescue efforts.

Typical Situation

- The ANI information is used for PSAP callback, which assumes that the ELINs are PSTN dialable numbers.
- The ELINs are PSTN numbers associated with the MLTS. If someone calls the ELIN from the PSTN, the call will terminate on an interface controlled by the MLTS.
- It is the responsibility of the MLTS system administrator to program the call routing so that calls made to any ELIN in the system will ring a phone (or multiple phones) in the immediate vicinity of the associated ERL.
- Once the ERL-to-ELIN mapping is established, it needs be modified only when there are changes to the physical situation of the enterprise. If phones are simply added, moved, or deleted from the system, the ERL-to-ELIN mapping and its associated ANI/ALI database records need not be changed.

Exceptional Situation

- Callback to the immediate vicinity of the originating ERL may be combined with (or even superseded by) routing the callback to an on-site emergency desk, which will assist the PSAP in reaching the original caller and/or provide additional assistance with the emergency situation at hand.
- The situation of the enterprise could change, for example, due to area code splits, city or county service changes requiring a new distribution of the public safety responsibilities, new buildings being added, or any other change that would affect the desired routing of a call for 911 purposes. Any of these events could require changes in the ERL-to-ELIN mapping and the ANI/ALI database records for the enterprise.

Dial Plan Considerations

It is highly desirable to configure a dial plan so that the system easily recognizes emergency calls, irrespective of whether an access code (for example, 9) is used or not. The emergency string for North America is generally 911. Cisco strongly recommends that you configure the system to recognize both the strings 911 and 9911.

Cisco also strongly recommends that you explicitly mark the emergency route patterns with Urgent Priority so that Unified CM does *not* wait for the inter-digit timeout (Timer T.302) before routing the call.

Other emergency call strings may be supported concurrently on your system. Cisco highly recommends that you provide your system users with training on the selected emergency call strings.

Also, it is highly desirable that users be trained to react appropriately if they dial the emergency string by mistake. In North America, 911 may be dialed in error by users trying to access a long distance number through the use of 9 as an access code. In such a case, the user should remain on the line to confirm that there is no emergency, and therefore no need to dispatch emergency personnel. Cisco Emergency Responder's on-site notification capabilities can help in identifying the phone at the origin of such spurious 911 calls by providing detailed accounts of all calls made to 911, including calls made by mistake. If the emergency dispatch center cannot confirm that a call to 911 was accidental, then emergency services must be dispatched to the calling location. More than three emergency services dispatches to a single customer in a month often times will result in a fine to the company.

In a multisite deployment, the dial plan configuration should ensure that the emergency calls are always routed through the PSTN gateway local to the site, thereby making sure that the emergency call is routed to the nearest PSAP within the jurisdiction. One of the mechanism to achieve this could be to use the Local Route Group feature of Cisco Unified CM. In the case of multisite deployments with centralized PSTN access, local call routing to the PSAP is not possible. For deployments with centralized PSTN access, the Unified CM administrator must verify that the PSTN provider will route emergency calls to the proper PSAP based on ANI or ELIN. If the service provider cannot provide emergency call routing services for multiple sites, then any site not included in E911 coverage must have a location connection (an analog line) or the centralized PSTN access must support 911 call delivery for remote sites (a SIP trunk). (See the examples in the section on Interface Type, page 15-7.)

Also, in a multisite deployment it is very important to make sure that the emergency number is always reachable and routed through the local PSTN gateway for the mobility users (extension mobility and device mobility) independent of the implemented Class of Service (CoS). If the site/device approach is being used, the device calling search space (CSS) could be used to route the emergency calls.

Cisco recommends enabling Calling Party Modification on Cisco Emergency Responder. When this feature is enabled, the calling party number is replaced with the ELIN by Cisco Emergency Responder for the emergency call. If Calling Party Modification is not enabled, either the DID will be sent to the PSAP or Cisco Unified CM must be configured to replace the calling party with the ELIN defined on the route pattern or the gateway.

Gateway Considerations

Consider the following factors when selecting the gateways to handle emergency calls for your system:

- Gateway Placement, page 15-17
- Gateway Blocking, page 15-17
- Answer Supervision, page 15-18
- Answer Supervision, page 15-18

Gateway Placement

Within the local exchange carrier (LEC) networks, 911 calls are routed over a locally significant infrastructure based on the origin of the call. The serving Class 5 switches are connected either directly to the relevant PSAP for their location or to an E911 selective router, which itself is connected to a group of PSAPs significant for its region.

With Cisco's IP-based enterprise communications architecture, it is possible to route calls on-net to gateways that are remotely situated. As an example, an endpoint located in San Francisco could have its calls carried over an IP network to a gateway situated in San Jose, and then sent to the LEC's network.

For 911 calls, it is critical to choose the egress point to the LEC network so that emergency calls are routed to the appropriate local PSAP. In the example above, a 911 call from the San Francisco endpoint, if routed to a San Jose gateway, could not reach the San Francisco PSAP because the San Jose LEC switch receiving the call does not have a link to the E911 selective router serving the San Francisco PSAP. Furthermore, the San Jose area 911 infrastructure would not be able to route the call based on a San Francisco calling party number.

As a general rule, route 911 calls to a gateway physically located with the originating endpoint. Contact the LEC to explore the possibility of using a common gateway to aggregate the 911 calls from multiple locations. Be aware that, even if the 911 network in a given region lends itself to using a centralized gateway for 911 calls, it might be preferable to rely on gateways located with the calling phones to prevent 911 call routing from being impacted during WAN failures.

Gateway Blocking

It is highly desirable to protect 911 calls from "all trunks busy" situations. If a 911 call needs to be connected, it should be allowed to proceed even if other types of calls are blocked due to lack of trunking resources. To provide for such situations, you can dedicate an explicit trunk group just for 911 calls.

It is acceptable to route emergency calls exclusively to an emergency trunk group. Another approach is to send emergency calls to the same trunk group as the regular PSTN calls (if the interface permits it), with an alternative path to a dedicated emergency trunk group. The latter approach allows for the most flexibility.

As an example, we can point emergency calls to a PRI trunk group, with an alternate path (reserved exclusively for emergency calls) to POTS lines for overflow conditions. If we put 2 POTS lines in the alternate trunk group, we are guaranteeing that a minimum of two simultaneous 911 calls can be routed, in addition to any calls that were allowed in the main trunk group.

If the preferred gateway becomes unavailable, it may be acceptable to overflow emergency calls to an alternate number so that an alternate gateway is used. For example, in North America calls dialed as 911 could overflow to an E.164 (non-911) local emergency number. This approach does not take advantage of the North American 911 network infrastructure (that is, there is no selective routing, ANI, or ALI services), and it should be used only if it is acceptable to the applicable public safety authorities and only as a last resort to avoid rejecting the emergency call due to a lack of network resources.

Answer Supervision

Under normal conditions, calls made to an emergency number should return answer supervision upon connection to the PSAP. The answer supervision may, as with any other call, trigger the full-duplex audio connection between the on-net caller and the egress interface to the LEC's network.

With some North American LECs, answer supervision might not be returned when a "free" call is placed. This may be the case for some toll-free numbers (for example, 800 numbers). In exceptional situations, because emergency calls are considered "free" calls, answer supervision might not be returned upon connection to the PSAP. You can detect this situation simply by making a 911 test call. Upon connection to the PSAP, if audio is present, the call timer should record the duration of the ongoing call; if the call timer is absent, it is very likely that answer supervision was not returned. If answer supervision is not returned, Cisco highly recommends that you contact the LEC and report this situation because it is most likely not the desired functionality.

If this situation cannot be rectified by the Local Exchange Carrier, it would be advisable to configure the egress gateway *not* to require answer supervision when calls are placed to the LEC's network, and to cut through the audio in both directions so that progress indicator tones, intercept messages, and communications with the PSAP are possible even if answer supervision is not returned.

By default, Cisco IOS-based H.323 gateways must receive answer supervision in order to connect audio in both directions. To forego the need for answer supervision on these gateways, use the following commands:

- **progress_ind alert enable 8**

  This command provides the equivalent of receiving a progress indicator value of 8 (in-band information now available) when alerting is received. This command allows the POTS side of the gateway to connect audio toward the origin of the call.

- **voice rtp send-recv**

  This command allows audio cut-through in both backward and forward directions before a connect message is received from the destination switch. This command affects all Voice over IP (VoIP) calls when it is enabled.

Be advised that, in situations where answer supervision is not provided, the call detail records (CDRs) will not accurately reflect the connect time or duration of 911 calls. This inaccuracy can impede the ability of a call reporting system to document the relevant statistics properly for 911 calls.

In all cases, Cisco highly recommends that you test 911 call functionality from all call paths and verify that answer supervision is returned upon connection to the PSAP.

Cisco Emergency Responder Design Considerations

Device mobility brings about special design considerations for emergency calls. Cisco Emergency Responder (Emergency Responder) can be used to track device mobility and to adapt the system's routing of emergency calls based on a device's dynamic physical location.

Device Mobility Across Call Admission Control Locations

In a centralized call processing deployment, Cisco Emergency Responder can detect Cisco endpoint relocation and reassign relocated endpoints to appropriate ERLs automatically. However, Cisco Unified CM location-based call admission control for a relocated endpoint might not properly account for the WAN bandwidth usage of the phone in the new location, yielding possible over-subscription or under-subscription of WAN bandwidth resources. For example, if you physically move a phone from Branch A to Branch B, the endpoint's call admission control location remains the same (Location_A), and it is possible that calls made to 911 from that endpoint would be blocked due to call admission control denial if all available bandwidth to Location_A is in use for other calls. To avoid such blocking of calls, manual intervention might be required to adapt the device's location and region parameters.

Cisco Unified CM device mobility provides a way to update the endpoint's configuration automatically (including its calling search space and location information) in Unified CM to reflect its new physical location. If device mobility is not used, manual configuration changes may be necessary in Cisco Unified CM.

For more details on the Device Mobility feature, refer to the section on Device Mobility, page 21-15.

Default Emergency Response Location

If Cisco Emergency Responder cannot directly determine the physical location of an endpoint, it assigns a default emergency response location (ERL) to the call. The default ERL points all such calls to a specific PSAP. Although there is no universal recommendation as to where calls should be sent when this situation occurs, it is usually desirable to choose a PSAP that is centrally located and that offers the largest public safety jurisdiction. It is also advisable to populate the ALI records of the default ERL's emergency location identification numbers (ELINs) with contact information for the enterprise's emergency numbers and to offer information about the uncertainty of the caller's location. In addition, it is advisable to mark those ALI records with a note that a default routing of the emergency call has occurred. Alternatively, the call may be routed to an internal security office to determine the caller's location.

Cisco Emergency Responder and Location Awareness for Wireless Clients

Cisco Emergency Responder 11.5 and later releases can track wireless endpoints and clients to an access point in the enterprise. To minimize configuration changes in Cisco Emergency Responder, all access points must be synchronized from Cisco Unified Communications Manager. The synchronization process also handles any access point additions, updates, or removals that occur in Cisco Unified CM. Any access point changes in Unified CM are seen in Emergency Responder within 2 minutes of the change. Access points cannot be defined within Emergency Responder, and all access points that are to be used for location identification in Emergency Responder must be defined in Unified CM. For access point management, Unified CM uses the Cisco Wireless LAN Controller Synchronization Service to automatically synchronize access points into the Unified CM database. The Cisco Wireless LAN Controller Synchronization Service integrates with Cisco Wireless LAN Controllers (WLCs) for access point information. If another vendor is used for WLC services, then the access points must be bulk imported into the Cisco Unified CM database using the Bulk Administration Tool (BAT).

For a mobile client or wireless device to be associated to a wireless access point, the client or device must send the Basic Service Set Identifier (BSSID) of the associated access point to Cisco Unified CM. Due to the frequency of updates that a mobile client can generate, Unified CM limits the rate of location updates from mobile devices and wireless clients to 90 updates per second per node. If location updates exceed this rate for a sustained period of time, Unified CM defers further updates with a 480 "Busy Here" message. The client responds by waiting a period of time before sending the location update again. The amount of delay before sending the update again dependents on the client and not on Cisco Emergency Responder or Unified CM.

When a mobile client or wireless device updates its location in Cisco Unified CM, the update is reflected in Cisco Emergency Responder in less than 2 minutes.

Cisco Emergency Responder and Extension Mobility

Cisco Emergency Responder supports Extension Mobility within a Cisco Unified CM cluster. It can also support Extension Mobility Cross-Cluster (EMCC), provided that both Cisco Unified CM clusters are supported either by a common Cisco Emergency Responder server or group, or by two Cisco Emergency Responder servers or groups configured as a Cisco Emergency Responder cluster. In either case, the Cisco Unified CM clusters must not be configured to use the Adjunct Calling Search Space (CSS) associated with EMCC for 911 calls, but must be configured to use Cisco Emergency Responder for all 911 calls in both Cisco Unified CM clusters.

Cisco Emergency Responder and Video

Cisco Emergency Responder can discover Cisco Video Collaboration endpoints in the following ways, depending on their capabilities:

- Video Collaboration Endpoints that Support CDP, page 15-20
- Video Collaboration Endpoints that Do Not Support CDP, page 15-21

Regardless of which way the video endpoints are discovered, it is important to note that video is not supported as media for emergency calling to the PSAP.

Note The topics discussed in this chapter apply to Cisco Emergency Responder only when it is used in conjunction with Cisco Unified Communications Manager (Unified CM). Cisco TelePresence Video Communication Server (VCS) currently does not support emergency services.

Video Collaboration Endpoints that Support CDP

For video collaboration endpoints that support Cisco Discovery Protocol (CDP) and that are within the corporate premises, Cisco recommends treating them like any other collaboration endpoints tracked by Cisco Emergency Responder through CDP, as described by the Emergency Responder switch configuration information in the latest version of the *Cisco Emergency Responder Administration Guide*, available at

http://www.cisco.com/en/US/partner/products/sw/voicesw/ps842/prod_maintenance_guides_list.html

For video collaboration endpoints with CDP support that are outside the corporate premises, Cisco recommends treating them like voice collaboration endpoints as described in the information for off-premises support of IP phones in the latest version of the *Off-Premise Location Management User Guide for Cisco Emergency Responder*, available at

http://www.cisco.com/en/US/partner/products/sw/voicesw/ps842/products_user_guide_list.html

Video Collaboration Endpoints that Do Not Support CDP

For video collaboration endpoints that do not support Cisco Discovery Protocol (CDP), Cisco recommends using a dedicated line for a voice collaboration endpoint. If you require tracking of the video collaboration endpoint, then Cisco recommends configuring an IP subnet ERL as described in the information about setting up IP subnet-based ERLs found in the latest version of the *Cisco Emergency Responder Administration Guide,* available at

http://www.cisco.com/en/US/partner/products/sw/voicesw/ps842/prod_maintenance_guides_list.html

Cisco Emergency Responder and Off-Premises Endpoints

In cases where endpoints are located outside of the enterprise boundary but connect back to the enterprise using VPN or VPN-less solutions (for example, Cisco Expressway mobile and remote access from a home office or hotel), Cisco Emergency Responder will not be able to determine the location of the caller. Furthermore, it is unlikely that the system would have a gateway properly situated to allow sending the call to the appropriate PSAP for the caller's location.

It is a matter of enterprise policy to allow or not to allow the use of off-premises endpoints for 911 calls through the enterprise. It might be advisable to disallow 911 calls by policy for those endpoints that connect over the Internet through VPN or Cisco Expressway. Nevertheless, if such a user were to call 911, the best-effort system response would be to route the call to either an on-site security force or a large PSAP close to the system's main site.

The following paragraph is an example notice that you could issue to users to warn them that emergency call functionality is not guaranteed for off-premises endpoints and users:

> *Emergency calls should be placed from devices that are located at the site for which they are configured (for example, your office). A local safety authority might not answer an emergency call placed from a device that has been removed from its configured site. If you must use this device for emergency calls while away from your configured site, be prepared to provide the answering public safety authority with specific information regarding your location. Use a device that is locally configured to the site (for example, your hotel phone or your home phone) for emergency calls when traveling or telecommuting.*

Cisco Emergency Responder also supports integration with Intrado V9-1-1, an emergency call delivery service that can reach almost any PSAP in the United States. With the combination of Cisco Emergency Responder and Intrado V9-1-1, users of IP phones and softphones outside the enterprise can update their locations by using the display screen on most Cisco IP Phones and Cisco IP Communicator or by using a web page provided by Cisco Emergency Responder. Emergency calls from an off-premises location will then be delivered through Cisco Emergency Responder to Intrado and then to the appropriate PSAP for the caller's location.

Test Calls

For any enterprise telephony system, it is a good idea to test 911 call functionality, not only after the initial installation, but regularly, as a preventive measure.

The following suggestions can help you carry out the testing:

- Contact the PSAP to ask for permission before doing any tests, and provide them with the contact information of the individuals making the tests.
- During each call, indicate that it is *not* an actual emergency, just a test.
- Confirm the ANI and ALI that the call taker has on their screen.

- Confirm the PSAP to which the call was routed.
- Confirm that answer supervision was received by looking at the call duration timer on the endpoint. An active call timer is an indication that answer supervision is working properly.

PSAP Callback to Shared Directory Numbers

Cisco Emergency Responder handles the routing of inbound calls made to emergency location identification numbers (ELINs). In cases where the line from which a 911 call was made is a shared directory number, the PSAP callback will cause all shared directory number appearances to ring. Any of the shared appearances can then answer the call, which means that it may not be the phone from which the 911 call originated.

In Cisco Unified CM 11.5 and later releases, a PSAP callback to a shared DN will ring only the device that placed the call to the PSAP. Unified CM will override device and line settings (such as Call Forward All and Do Not Disturb) to deliver the callback from emergency services.

Cisco Emergency Responder Deployment Models

Enterprise communications systems based on multiple Unified CM clusters can benefit from the functionality of Cisco Emergency Responder (Emergency Responder).

The *Cisco Emergency Responder Administration Guide* provides detailed descriptions of the terms used herein, as well as the background information required to support the following discussion. Of specific interest is the chapter on *Planning for Cisco Emergency Responder.* This documentation is available at

http://www.cisco.com/en/US/products/sw/voicesw/ps842/prod_maintenance_guides_list.html

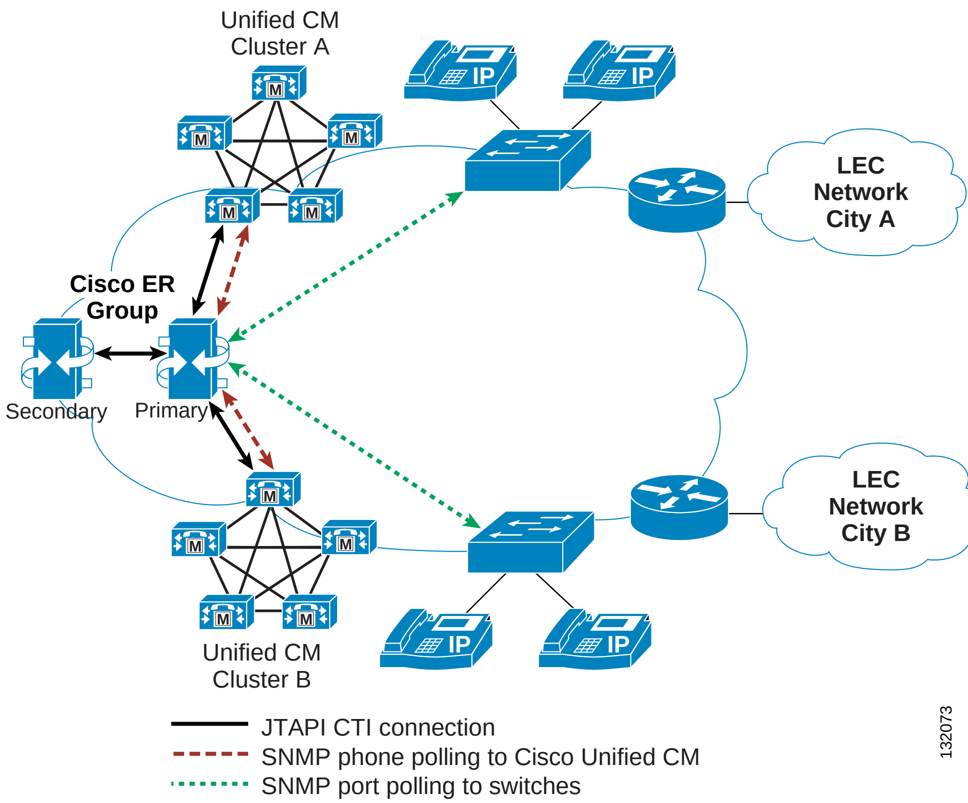
Note Cisco Emergency Responder does not support Cisco Unified Communications Manager Express (Unified CME) or Survivable Remote Site Telephony (SRST). In case of SRST deployment, configure the appropriate dial-peer to route the 911 calls to the PSTN with the published site number. Unified CME natively supports E911.

Single Cisco Emergency Responder Group

A single Emergency Responder group can be deployed to handle emergency calls from two or more Unified CM clusters. The design goal is to ensure that an emergency call from any phone is routed to the Cisco Emergency Responder group, which will assign an ELIN and route the call to the appropriate gateway based on the endpoint's location.

One advantage of using a single Cisco Emergency Responder group is that all ERLs and ELINs are configured into a single system. An endpoint registered on any cluster will be located by the single Cisco Emergency Responder group because that group is responsible for polling all of the system's access switches. Figure 15-3 illustrates a single Cisco Emergency Responder group interfaced with two Unified CM clusters.

Figure 15-3 A Single Cisco Emergency Responder Group Connected to Two Unified CM Clusters



The single Cisco Emergency Responder group in Figure 15-3 interfaces with the following components:

- Each Unified CM cluster, via SNMP, to collect information about their respective configured endpoints.
- Enterprise access switches, via SNMP, where IP telephony endpoints are connected. This connection is not required if the endpoint locations are being identified based on IP subnets. For details on configuring IP subnet-based ERLs, refer to the *Cisco Emergency Responder Configuration* chapter in the *Cisco Emergency Responder Administration Guide*, available at

  http://www.cisco.com/en/US/products/sw/voicesw/ps842/prod_maintenance_guides_list.html
- Each Unified CM cluster, via JTAPI, to allow for the call processing required by any endpoint that dials 911 – for example, identification of the calling endpoint's ERL, assignment of the ELIN, redirection of the call to the proper gateway (based on the calling endpoint's location), and the handling of the PSAP callback functionality.
- Each Unified CM cluster, via SNMP, to collect access point information from a Cisco Wireless LAN Controller (WLC).

The version of the JTAPI interface used by Cisco Emergency Responder is determined by the version of the Unified CM software to which it is connected. At system initialization, Cisco Emergency Responder interrogates the Unified CM cluster and loads the appropriate JTAPI Telephony Service Provider (TSP). Because there can be only one version of JTAPI TSP on the Cisco Emergency Responder server, all Unified CM clusters to which a single Cisco Emergency Responder group is interfaced *must* run the same version of Unified CM software.

For some deployments, this software version requirement might present some difficulties. For instance, during a Unified CM upgrade, different clusters will be running different versions of software, and some of the clusters will be running a version of JTAPI that is not compatible with the version running on the Cisco Emergency Responder servers. When this situation occurs, emergency calls from the cluster running a version of JTAPI different than that of the Cisco Emergency Responder group might receive the call treatment provided by the call forward settings of the emergency number's CTI Route Point.

When considering if a single Cisco Emergency Responder group is appropriate for multiple Unified CM clusters, apply the following guidelines:

- Make Unified CM upgrades during an acceptable maintenance window when emergency call volumes are as low as possible (for example, after hours, when system use is at a minimum).
- Use a single Cisco Emergency Responder group only if the quantity and size of the clusters allow for minimizing the amount of time when dissimilar versions of JTAPI are in use during software upgrades.

For example, a deployment with one large eight-server cluster in parallel with a small two-server cluster could be considered for use with a single Cisco Emergency Responder group. In this case, it would be best to upgrade the large cluster first, thus minimizing the number of users (those served by the small cluster) that might be without Cisco Emergency Responder service during the maintenance window of the upgrade. Furthermore, the small cluster's users can more appropriately be served by the temporary static routing of emergency calls in effect while Cisco Emergency Responder is not reachable because they can be identified by the single ERL/ELIN assigned to all non-ER calls made during that time.

Multiple Cisco Emergency Responder Groups

Multiple Cisco Emergency Responder groups can also be deployed to support multi-cluster systems. In this case, each ER group interfaces with the following components:

- A Unified CM cluster via the following methods:
  - SNMP, to collect information about its configured endpoints
  - JTAPI, to allow for the call processing associated with redirection of the call to the proper gateway or, in the case of roaming endpoints, the proper Unified CM cluster
- The access switches (via SNMP) to which most of the endpoints associated with the Unified CM of the Cisco Emergency Responder group are most likely to be connected
- Each Unified CM cluster (via SNMP) to collect Access Point information from a Cisco Wireless LAN Controller (WLC)

This approach allows Unified CM clusters to run different versions of software because each is interfaced to a separate Cisco Emergency Responder group.

To allow endpoints to roam between various parts of the network and still be tracked by Cisco Emergency Responder, you might have to configure the Cisco Emergency Responder groups into a Cisco Emergency Responder cluster. For details on Cisco Emergency Responder clusters and groups, refer to the chapter on *Planning for Cisco Emergency Responder* in the *Cisco Emergency Responder Administration Guide*, available at

http://www.cisco.com/en/US/products/sw/voicesw/ps842/prod_maintenance_guides_list.html

Figure 15-4 presents a sample topology illustrating some of the basic concepts behind Cisco Emergency Responder clustering.

Figure 15-4 Multiple Cisco Emergency Responder Groups

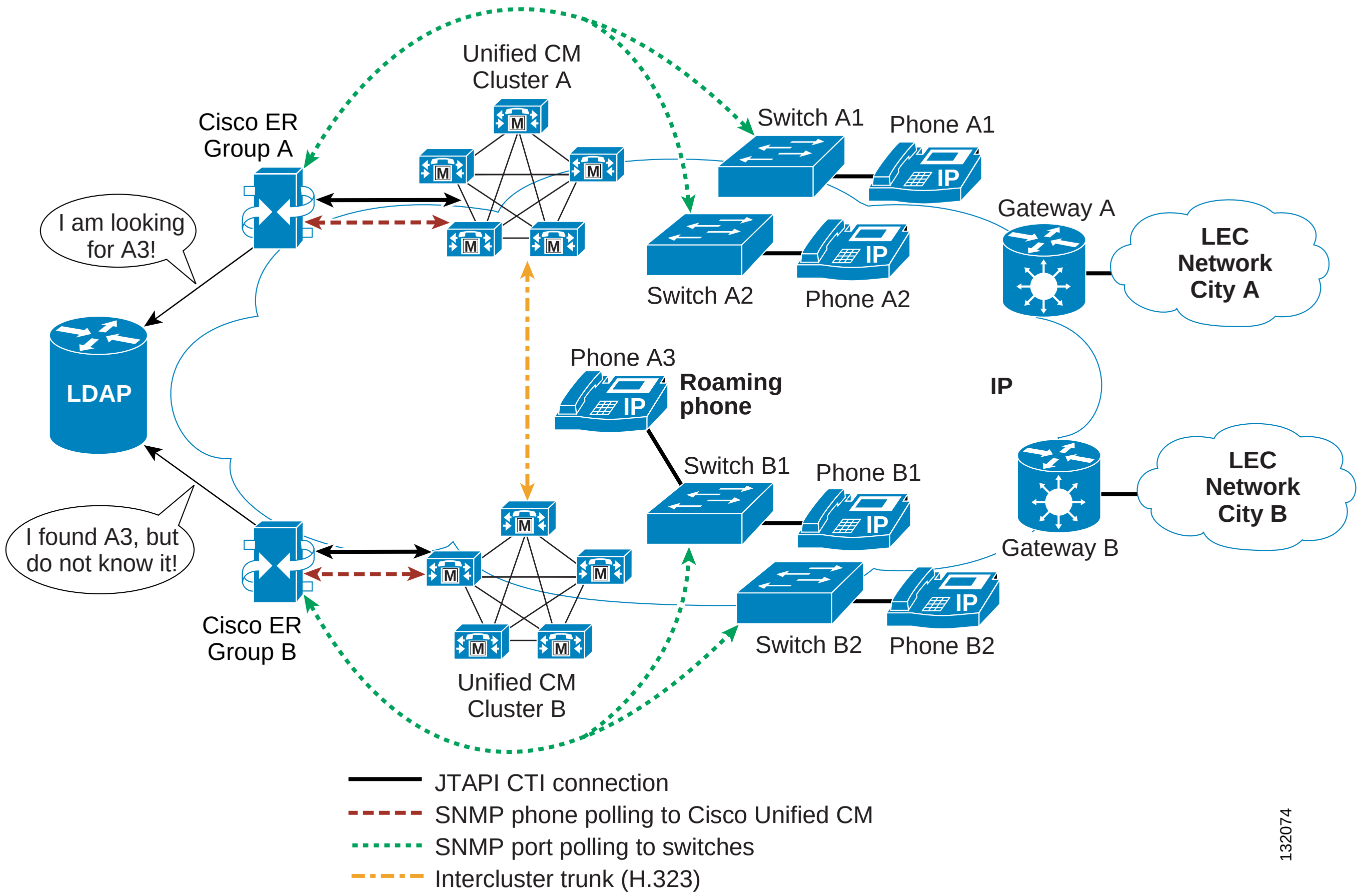


Figure 15-4 illustrates the following topology:

- Cisco Emergency Responder group A is interfaced to Unified CM cluster A to access switches A1 and A2, and it is deemed to be the home Cisco Emergency Responder group of all endpoints registered to Unified CM cluster A.
- Likewise, Cisco Emergency Responder group B is interfaced to Unified CM cluster B to access switches B1 and B2, and it is deemed to be the home Cisco Emergency Responder group of all endpoints registered to Unified CM cluster B.

Endpoint Movements Within the Tracking Domain of a Cisco Emergency Responder Group

The emergency call processing for endpoints moving between access switches controlled by the same home Cisco Emergency Responder group is the same as the processing done for a deployment with a single Unified CM cluster. For example, an endpoint moving between access switches A1 and A2 remains registered with Unified CM cluster A, and its location is determined by Cisco Emergency Responder group A both before and after the move. The endpoint is still under full control of Cisco Emergency Responder group A, for both the discovery of the endpoint by Unified CM cluster A and the determination of the endpoint's location on switch A2 by Cisco Emergency Responder. The endpoint is therefore not considered to be an unlocated phone.

Endpoint Movements Between the Various Tracking Domains of a Cisco Emergency Responder Cluster

A Cisco Emergency Responder cluster is essentially a collection of Cisco Emergency Responder groups that share location information. Each group shares the location of any endpoint it finds on an access switch or in an IP subnet.

Cisco Emergency Responder groups also share information about endpoints that cannot be located within a Cisco Emergency Responder group's tracking domain (in switches or IP subnets) but which are known to be registered in the group's associated Unified CM cluster. Such endpoints are deemed *unlocated*.

If an endpoint is roaming between access switches monitored by different Cisco Emergency Responder groups, those groups must be configured in a Cisco Emergency Responder cluster so they can exchange information about the endpoint's location. For example, endpoint A3 is registered with Unified CM cluster A, but it is connected to an access switch controlled by Cisco Emergency Responder group B. Cisco Emergency Responder group A is aware that endpoint A3 is registered with Unified CM cluster A, but group A cannot locate endpoint A3 in any of the site A switches. Therefore, endpoint A3 is deemed *unlocated* by Cisco Emergency Responder group A.

Cisco Emergency Responder group B, on the other hand, has detected the presence of endpoint A3 in one of the switches that it monitors. Because the endpoint is not registered with Unified CM cluster B, endpoint A3 is advertised through the Cisco Emergency Responder database as an *unknown* endpoint.

Because the two Cisco Emergency Responder groups are communicating through a replicated database table, they can determine that Cisco Emergency Responder group B's *unknown* endpoint A3 is the same as Cisco Emergency Responder group A's *unlocated* endpoint A3.

The Unlocated Phone page in Cisco Emergency Responder group A will display the endpoint's MAC address along with the remote Cisco Emergency Responder group (in this, case Cisco Emergency Responder group B).

Emergency Call Routing within a Cisco Emergency Responder Cluster

Cisco Emergency Responder clustering also relies on route patterns that allow emergency calls to be redirected between pairs consisting of a Unified CM cluster and a Cisco Emergency Responder. For more details, refer to the section on *Creating Route Patterns for Inter-Cisco Emergency Responder Group Communications* in the *Cisco Emergency Responder Administration Guide*, available at

http://www.cisco.com/en/US/products/sw/voicesw/ps842/prod_maintenance_guides_list.html

If endpoint A3 places an emergency call, the call signaling flow will be as follows:

1. Endpoint A3 sends the emergency call string to Unified CM cluster A for processing.
2. Unified CM cluster A sends the call to Cisco Emergency Responder group A for redirection.
3. Cisco Emergency Responder group A determines that endpoint A3 is located in Cisco Emergency Responder group B's tracking domain, so it redirects the call to a route pattern that points to Unified CM cluster B.
4. Unified CM cluster A sends the call to Unified CM cluster B over a SIP trunk or an intercluster trunk.
5. Unified CM cluster B sends the call to Cisco Emergency Responder group B for redirection.
6. Cisco Emergency Responder group B identifies the ERL and ELIN associated with endpoint A3's location (based on calling party number) and redirects the call to Unified CM cluster B. The calling number is transformed into the ELIN associated with the ERL of endpoint A3, and the called number is modified to route the call to the proper gateway.

7. Unified CM cluster B routes the call according to the new called number information obtained from Cisco Emergency Responder group B.
8. Unified CM cluster B sends the call out the gateway toward the Emergency PSTN network.

Note The ERL and ELIN match in step 6 is based on the calling party number of the endpoint placing the call to 911. If the SIP trunk or intercluster trunk modifies the calling party number (perhaps to a full +E.164 number), then the Emergency Responder for Group B will not be able to match the calling party number over the trunk with the directory number learned from the access switch Cisco Discovery Protocol (CDP) neighbor. Therefore, emergency calls that traverse a SIP trunk or intercluster trunk must not undergo any calling party transformations.

WAN Deployment of Cisco Emergency Responder

Cisco Emergency Responder supports two main sites using clustering over the WAN. Install one Emergency Responder server in each site, and configure one server as the publisher and the other server as a subscriber. The Emergency Responder publisher should be located with the primary Unified CM CTI Manager, and the Emergency Responder subscriber should be located with the secondary Unified CM CTI Manager. Any Unified CM server remote from either Emergency Responder server must be within 80 ms round-trip time (RTT) of both Emergency Responder servers. The Emergency Responder publisher and subscriber must also be within 80 ms RTT of each other. The minimum bandwidth required between the Cisco Emergency Responder servers is 1.544 Mbps.

Emergency Call Routing Using Unified CM Native Emergency Call Routing

Customers that require accurate location identification but have a single site or small number of locations that need to be identified, can use the Cisco Unified Communications Manager Native Emergency Call Routing feature. The Native Emergency Call Routing feature allows an administrator to define Emergency Location Identification Numbers (ELINs) at the device pool level or device level so that a device's location can be determined and identified at the public safety answering point (PSAP).

Cisco Unified CM Native Emergency Call Routing provides the following functionality:

- ELIN association based on a static device assignment or device pool assignment
- Dynamic association of the ELIN to the calling phone for callback purposes
- For mobile devices, Device Mobility Groups used to track mobile devices with Native Emergency Call Routing
- Automatic replacement of the calling party number with the appropriate ELIN
- Routing emergency calls to the appropriate gateway for emergency call completion

Design Considerations for 911 Native Emergency Call Routing Services

When designing an emergency call routing plan using Cisco Unified CM Native Emergency Call Routing services, give special consideration to the boundaries of an emergency location inside a building. An emergency location should be an identifiable location with physical or logical boundaries to reduce the amount of time for emergency services to locate an individual in an emergency situation. Examples of physical or logical boundaries can include: a single floor of a building, a lab, an office, or a directional floor indicator (for example, West side of first floor).

The design for Native Emergency Call Routing requires an ELIN to be defined and assigned to devices or devices pools, but the Native Emergency Call Routing feature does not allow the administrator to define the ERL information to be associated with the ELIN. The ERL definition for a given ELIN must be done outside of Cisco Unified CM and uploaded to the local PSAP per the instructions provided by the local exchange carrier when establishing E911 services.

Similar to a Cisco Emergency Responder deployment, Native Emergency Call Routing can support multiple unique and concurrent calls to emergency services from the same location. Native Emergency Call Routing allows the creation of a pool of ELINs that are associated with an emergency location. The number of locations that can be defined is based on the number of ELINs assigned to an individual Emergency Location (ELIN) Group. Native Emergency Call Routing supports a maximum of 100 ELINs. If the deployment requires only one concurrent call per a location, then the system can support 100 unique Emergency Location Groups. If the deployment requires the ability to track 2 concurrent callers from the same location, then the administrator must define 2 ELINs for a single Emergency Location (ELIN) Group. If 2 ELINs are required for a single location, Unified CM will be able to support 50 locations (2 ELINs * 50 ERLs = 100 ELINs). Using more ELINs to support concurrent and uniquely identified callers from a location will reduce the total number of locations that can be defined. The following formula can be used to determine the maximum number of locations that can be defined based on the number of concurrent and unique callers from an ERL:

100/(Number of unique and concurrent callers per ERL) = Max ERLs

ELINs are not required to be the same for each Emergency Location (ELIN) Group. If one ERL covers a high-density user population, the Emergency Location (ELIN) Group may contain 4 ELINs to support 4 concurrent and unique emergency callers. But if the same building has a large lab floor or warehouse that has a small number of regular employees, then that location might have only one ELIN assigned to the Emergency Location (ELIN) Group.

If the PSAP needs to call back and get additional information from the caller, the call will return to Unified CM using the ELIN that originated the call. To route the return call correctly, the dial plan must be configured so that the inbound called number matches the ELIN defined in Unified CM. If the inbound trunk delivers only the last 5 digits of the called party, then the administrator must include a translation pattern to expand the collected digits to match the ELIN. For proper return call operation, the called number must match exactly the ELIN number as defined in Unified CM. Although ELINs can be any number in a customer's DID range, Cisco recommends keeping the ELIN numbers contiguous to use as few call translation patterns as possible.

ALI Formats

In multi-cluster configurations, there might be instances where the physical locations of ERLs and ELINs defined in a single Cisco Emergency Responder group span the territory of more than one phone company. This condition can lead to situations where records destined for different phone companies have to be extracted from a common file that contains records for multiple LECs.

Cisco Emergency Responder exports this information in ALI records that conform to National Emergency Number Association (NENA) 2.0, 2.1, and 3.0 formats. However, many service providers do not use NENA standards. In such cases, you can use the ALI Formatting Tool (AFT) to modify the ALI records generated by Cisco Emergency Responder so that they conform to the formats specified by the service provider. The service provider can then use the reformatted file to update their ALI database.

The ALI Formatting Tool (AFT) enables you to perform the following functions:

- Select a record and update the values of the ALI fields. AFT allows you to edit the ALI fields to customize them to meet the requirements of various service providers. The service provider can then read the reformatted ALI files and use them to update their ELIN records.
- Perform bulk updates on multiple ALI records. Using the bulk update feature, you can apply common changes to all the records that you have selected.
- Selectively export ALI records based on area code, city code, or a four-digit directory number. By selecting to export all the ALI records in an area code, for example, you can quickly access all the ELIN records for each service provider, thereby easily supporting multiple service providers.

Given the flexibility of the AFT, a single Cisco Emergency Responder group can export ALI records in multiple ALI database formats. For a Cisco Emergency Responder group serving a Unified CM cluster with sites in the territories of two LECs, the basic approach is as follows:

1. Obtain an ALI record file output from Cisco Emergency Responder in standard NENA format. This file contains the records destined for multiple LECs.
2. Make a copy of the original file for each required ALI format (one copy per LEC).
3. Using the AFT of the first LEC (for example, LEC-A), load a copy of the NENA-formatted file and delete the records of all the ELINs associated with the other LECs. The information to delete can usually be identified by NPA (or area code).
4. Save the resulting file in the required ALI format for LEC-A, and name the file accordingly.
5. Repeat steps 3 and 4 for each LEC.

For more information about the ALI formatting tools, refer to the online documentation available at

http://www.cisco.com/en/US/products/sw/voicesw/ps842/prod_maintenance_guides_list.html

For LECs not listed at this URL, the output from Emergency Responder can be formatted using standard text file editing tools, such as spreadsheet programs and standard text editors.

Exhibit D

SIPTRUNK

January 13, 2016

How SIP Trunks Handle 911 Calls

One of the most important phone numbers that you'll ever dial (besides calling your mother – which you should do once a week) is 9-1-1.

When your phone is connected to a landline, and is one that you can physically see extending from your house to the phone pole on the street, you just don't think about it much.

And why should you? It's going on 50 years now since [the first 911 call was made in the U.S. in 1968] by Senator Rankin Fite in Haleyville, Alabama. This occurred shortly after the FCC met with AT&T and Congress agreed to the proposal for a single emergency calling service and passed legislation.

On one hand we take it for granted that the call gets routed to the right place. On the other hand isn't it reasonable to expect that a 911 call over VoIP should function exactly the same way?

VoIP and 911

Give the FCC page, VoIP and 911 Service, a quick read and you might be left with the impression of "caveat emptor."

"Have a clear understanding of any limitations of your 911 service," says the FCC and they provide a list that advises VoIP users that calls may not connect or properly connect to the PSAP (public safety answering point), calls may not work when the internet or power is down, and customers need to proactively provide their location and address information to their VoIP provider for 911 service to work properly.

However, in 2005 the FCC imposed VoIP regulations stating that, "providers of interconnected VoIP telephone services using the Public Switched Telephone Network (PSTN) meet Enhanced 911 (E911) obligations."

As such, all customers who are using SIPTRUNK.com services as their primary residential or business telephone carrier must activate 911 Emergency Services on at least one of their DIDs.

Click here to read the full SIPTRUNK.com e911 terms and conditions.

911 Calls Over SIP Trunks When All Channels are Busy

Even when you're using all of concurrent calling capacity some SIP trunking providers will give priority to a 911 or e911 call and let the call go through.

With other SIP trunks, the call will not go through if all channels are being used.

So on SIPTRUNK.com you'll need to pick at least 1 enhanced DID per physical location. If you have a big building, for example, you'll want to have one enhanced DID per floor.

Harpreet created a demo SIP Trunk Reseller account.

Get Your Free Trial Account Now

ıcate which attaches the address information to the call.

ɔund route for 911 which will override any caller ID you have on the trunk because it detects that you're calling 911, and not a standard number like 867-5309.

We have instructions on how to setup an enhanced DID for e911 and how to test that the proper address and caller ID information is being displayed by dialing the test number of "933."

For clarification:

[box type="download"]**Standard DID** – The lowest cost DID at SIPTRUNK.com where NO address information is input into our system[/box]

[box type="download"]**Enhanced DID** – An upgraded DID at SIPTRUNK.com where address information is input into our system which can be forwarded to a PSAP[/box]

The enhanced DID is key because it carries the address information that *needs* to be relayed to the 911 contact center. Standard DIDs are not meant for 911 calls. SIPTRUNK.com gets charged $75 for dialing 911 using a non-provisioned callerID number by our carrier – this charge then gets passed on to our customers.

In short, for the best 911 service on SIP trunks, buy at least one enhanced DID and in the long run you'll save time, money, and lives.

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Ready to Start SIP Trunking?

SIPTRUNK is the ideal SIP trunking provider for agents, dealers, VARs, manufacturers, distributors, master agents, and IT consultants looking to build a monthly recurring revenue stream selling SIP trunks.

Get Started

SIPTRUNK

At SIPTRUNK we provide a services platform designed for companies who want to build a SIP trunking practice and a recurring revenue stream selling SIP trunking services.

CHECK OUT WHAT PEOPLE SAYS

RESOURCES

- Certified PBX Platforms
- SIP Trunking Features
- SIP Trunking Basics
- What is SIP Trunking
- History of SIP Trunking
- The Benefits of SIP Trunking
- Explaining SIP Trunking to Your Customers
- Identifying SIP Trunking Opportunities
- Creating a Winning SIP Trunking Proposal

SIPTRUNK COMPATIBILITY

- SIP Trunks for 3CX
- SIP Trunks for AudioCodes
- SIP Trunks for Asterisk
- SIP Trunks for Avaya
- SIP Trunks for Elastix

GET STARTED

- How to Get Started
- About us
- Blog
- Privacy Policy
- Acceptable Use Policy
- Terms of Service

PRODUCTS

- Why Partner With Us?
- Features
- Pricing
- FAQs

COMPANY

- Sign In
- Get Started
- Contact
- Support

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Exhibit E

Vonage Business Plus

Business Plus SIP Trunking

Connect your Asterisk, TrixBox, or other IP PBX directly to the largest VoIP carrier in the US. You made the effort to deploy Asterisk in your Enterprise. Why limit yourself to PSTN Connectivity? Forget ATAs and Analog PSTN cards.

For hosted PBX from Vonage, see Vonage Business Solutions Reviews.

Vonage Business Plus plans allow direct SIP trunking from your VoIP Gateway, Asterisk/TrixBox, or other IP PBX directly to Vonage with unlimited simultaneous incoming and outgoing calls.

Vonage Business Plus service is only marketed, sold, and supported through authorized Vonage Resellers. The Business Plus plans from Vonage are the ideal solution for small to medium-sized enterprises requiring multiple phone lines. Whether your business is considering an upgrade of its current phone system or deployment of a next-generation IP-PBX, Vonage can provide your firm with the technology to take your telecommunications to the next level.

Standard Vonage service plans require an analogue telephone adapter (ATA) for each telephone line. While convenient for residential or home office use, this requirement can quickly become restrictive or cumbersome for businesses that require multiple phone lines or who have premise-based PBX or IP-PBX phone systems. The Vonage Business Plus plans allow customers to connect their SIP-based phones, IP-PBX and conventional PBX (using VoIP gateways) directly to the Vonage Broadband Phone network.

Vonage Business Plus are pay-per-minute plans that do not limit the number of simultaneous calls incoming or outgoing to your SIP trunk. You are only limited by your bandwidth and IP PBX configuration. Enjoy the freedom to use your SIP connection as it was intended. **Incoming calls are free, and Vonage-to-Vonage calls do not use minutes from your plan.**

Vonage Business Plus plans offer the same great Local Number Portability (LNP) features as standard Vonage plans. If LNP is offered in your area for standard Vonage service, then it is available for Vonage Business Plus.

From the administrative perspective, Vonage Business Plus accounts look like standard Vonage accounts. To administer a Business Plus account, simply log in to the standard Vonage web portal. However, instead of having one or two phone numbers, you will have 4 or more DIDs. In addition, a Vonage Business Plus account has an extra “Installation” page which includes the SIP registration information for each DID associate with the account. All other features remain the same, including 911, backup PSTN failover routing, and voice mail (which should be turned off when using an IP PBX).

Upgrading to Vonage Business Plus from Vonage standard service? No problem. Standard Vonage account phone numbers can be ported to Vonage Business Plus.

Currently, only US DIDs are available for your primary numbers; however, the Vonage Virtual Number feature can be used to obtain a virtual number in another area code (or country) and forward it to one of your primary DIDs. As well, Toll-Free numbers are also available.

For more information, visit The VoIP Connection.

Asterisk Vonage Configuration

The following configuration examples are used to configure your Asterisk PBX for Vonage service with a Vonage Business Plus account.

sip.conf
register= Phone_Number:Password@sphone1.vonage.net:5061

Phone_Number

Site Sponsor

Search...

Search

3CX PBX - F
Linux / Win /

Run on-premise on Windows / Applian

Or in YOUR cloud - Amazon, Azure

Integrated **iOS** & **A**

Zero admin and **bu** **security**

Brand new **web cli** **integrated web co**

ClicktoCall from ar

and many more

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Quick Links

Mobile VoIP

Business VoIP solutio

Hosted VoIP solution

Asterisk PBX downloa

Open Source VOIP So

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```
secret=Password
port=5061
nat=yes
insecure=very
host=sphone1.vonage.net
fromuser=Phone_Number
fromdomain=sphone1.vonage.net
dtmfmode=rfc2833
disallow=all
defaultexpirey=20
canreinvite=no
auth=md5
allow=g729
allow=ulaw
allow=alaw
```

- Be sure to turn off your Vonage Voice Mail feature for each DID in order to use your Asterisk voice mail system.

Support

Binary Systems, Inc., the creator of this Wiki page, provides complete setup and support for Vonage Business Plus service with Asterisk/TrixBox, Vegastream, AudioCodes, and other IP PBXs and VoIP gateways. ((Binary Systems, Inc.)) engineers integrated the first AudioCodes MediaPack equipment onto the Vonage network prior to the initial Vonage Business Plus rollout.

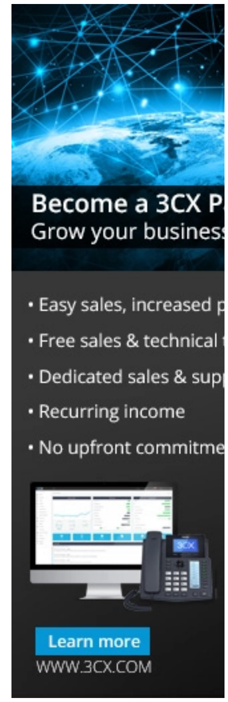
The VoIP Connection specializes in deploying Vonage Business Plus with Asterisk. We are a Digium Premier Solution Partner and a Digium Premier Reseller. Our Digium DCAP certified engineers have helped hundreds of businesses realize the benefits of connecting the worlds most flexible PBX to the worlds largest VoIP network.

Available from:

- Binary Systems, Inc.

See also

- Vonage for iPhone
- Vonage for BlackBerry
- Vonage



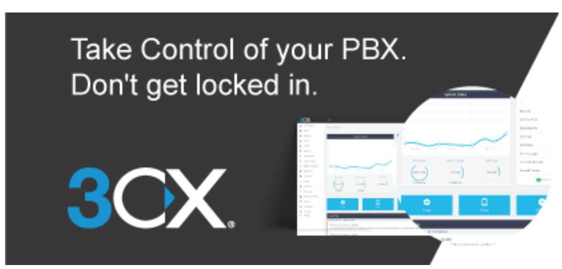


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Exhibit F

Build the future of communications.

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BY REBECCA MCKEE • 2017-10-11

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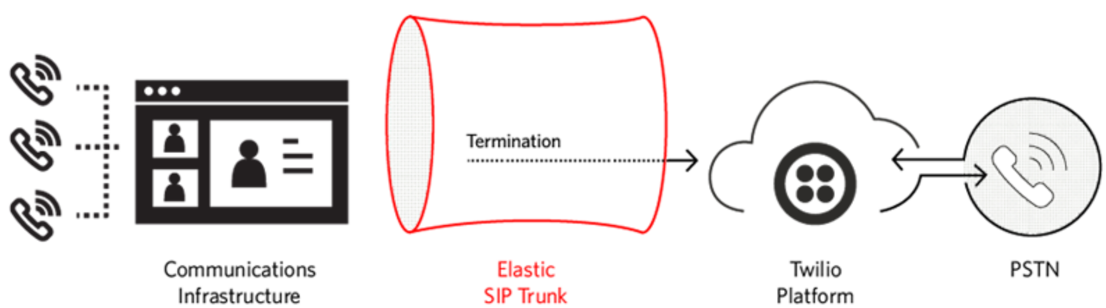
SIP Trunking Termination Calling is Now More Flexible with Master CPS



- Enhancement to Calls Per Second (CPS) feature for SIP Trunking termination.
- Save money and gain more control over capacity planning.
- Available in beta.

Today, we're excited to announce a huge enhancement to the Calls Per Second (CPS) feature for SIP Trunking termination. Master CPS is a feature that will save you money if using our traditional CPS feature today while allowing you to have more control over your capacity planning. It's a win-win!

What is CPS?

Before getting into all the exciting details of this enhancement, let's take a step back for a moment to understand what CPS is and how it relates to concurrency, a term more familiar amongst traditional SIP providers like AT&T and Verizon.

Concurrency refers to how many simultaneous calls can occur at same time. These traditional providers require you to plan for your highest peak traffic days, determine the highest number of concurrent calls that you may need to support, and pay for the number of "channels" that support that traffic for the duration of your contract term. Twilio, simply put, doesn't require *any* upfront capacity planning—we allow you to scale as needed, and you won't be charged for concurrency.

CPS refers to how many calls can be initiated in the same second. A business with 100 phone agents will need to be able to handle concurrency of 100 so that all their agents can be on the phone at the same time. However, they do not need CPS of 100, because it is extremely unlikely that all 100 agents would need to initiate calls during the same second. Twilio offers unlimited call concurrency, but since there is overhead in setting up a call, we do charge for CPS. Twilio will reserve extra outbound dialing capacity for users who have heavy dialing workloads.

Trunking CPS vs. Master CPS

There are two options for increasing Termination CPS for SIP Trunking: Trunking CPS and Master CPS.

An existing feature we offer is Trunk Level CPS, which allows you to provision each individual trunk with the CPS you require for each region. Pay per trunk, per region.

Master Account CPS is what we're really excited about! If you are currently using our traditional Trunk Level CPS capabilities, you'll have the opportunity to see your bill decrease! Master CPS allows you to provision your master account with the CPS you require for each region. Pay per region, per master account. The Master CPS settings are a pooled resource used by any trunk in the given account or corresponding sub-accounts. We also give you the ability to over-provision on the trunk level.

Would I Benefit From this New Feature?

Master CPS is the best fit for your needs if you have several trunks across multiple sub-accounts that need increased CPS, but you aren’t sure what level you’ll need on each of those trunks. For example, you have 4 trunks and based on your call volume, each trunk may require 2-6 CPS at a given time in the US1 region. To ensure successful calls are placed, you would want 6 CPS on each trunk to handle your highest peak times. You would be charged 4*6*$75 = $1,800. In reality, you probably aren’t initiating 6 calls per second ***all that often***, therefore, you’re paying a lot for something that used all that often.

With Master CPS, we’re giving you the ability to apply that 6 CPS at the master account level and only pay $75 flat vs $450. You can provision your trunks to whatever CPS level you’d like, but you’ll always be limited to 6 CPS since that’s what is set at the master account level.

How the Magic Happens

There are two “checkpoints” used to determine when calls will be allowed or rejected.

1. Actual calls on each trunk vs. Trunk CPS Limits: Do actual calls on a specific trunk exceed the CPS settings set on that trunk for a given region?
   - Yes – Calls start to be rejected based on the Trunk CPS settings
   - No – Go to next rule

2. Total actual calls across all trunks vs. Master Account CPS Limits: Do total actual calls across all trunks exceed the Master CPS settings in a given region?
   - Yes – Calls start to be rejected based on the master account CPS settings
   - No – All calls will go through

We have two debugger alerts set up in case calls are rejected. These alerts will tell you whether you hit a trunk limit or a master account limit. Then if you wish, you can change your CPS levels appropriately.

Here's an example of how this works:

Master Account CPS Setting:

North America – Oregon (US2): 25 CPS

	Trunk 1	Trunk 2	Trunk 3	Totals
Provisioned	20 CPS	25 CPS	30 CPS	75 CPS provisioned
Actual Calls	30 CPS	24 CPS	30 CPS	84 actual calls
Rule (1) Trunk Limits	Calls capped at 20	All calls allowed	All calls allowed	74 actual calls allowed
Rule (2) Master Limits	Cap all calls at 25 CPS because this is the max set at the master account level (calls from any trunk may be rejected)			

How to Get Started with Master CPS

If you'd like to start taking advantage of CPS, contact Sales and they will help you get your CPS levels modified.

We can't wait to see what you build!

AUTHORS | Rebecca McKee

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Exhibit G

Outbound Calling

Telzio enables you to make outbound calls from your Telzio phone number using the Telzio app, IP phones, and softphones.

Before you can get started with outbound calling, here are a few important notes:

- Telzio is an internet-based phone service, which means you can make calls from anywhere your device has internet connectivity.
- In order to make outbound calls, first you need to create a User for your phone system. Learn more about creating Users. (https://telzio.com/support/create-user)
- Telzio support unlimited concurrent calls, which means all your employees can make calls using a single Telzio phone number without callers experiencing a busy signal. With Telzio, there is no need for rollover lines.

IP Phones

Telzio works with a wide range of IP phones, including desk, wireless, and conference phones. You can browse our shop (https://telzio.com/shop) and order preconfigured phones that arrive ready for calling. If you need help choosing the right phone for your business, our supporters are happy to help. Contact us (https://telzio.com/contact) for recommendations.

In most cases, you can use existing IP phones with Telzio. You can search our Support (https://telzio.com/support) pages for step-by-step guides on configuring specific models to use with Telzio. It's important to note that our supporters can only provide limited support on phones that are not purchased directly from our shop.

Request Demo (/demo/) Start Free Trial (/trial/) Request a Quote (/quote)

Learn more about getting started with IP phones. (https://telzio.com/support/general-ip-phone-settings)

Mobile App

The Telzio mobile app enables you to start calling from day one, without investing in physical phones. The app provides similar functionality to an office phone, including:

- Hold
- Mute
- Speakerphone
- Transfer
- Call Recording
- Conferencing
- Speed Dial
- Contacts

From the app, you can call internally to other extensions, and transfer calls to other extensions. It's helpful to note that calling internally between extensions is included as part of your Telzio service, and does not count towards your usage.

Learn more about the mobile app. (https://telzio.com/support/set-up-telzios-mobile-softphone-app)

Softphones

A softphone is a third-party application you can download on your computer to make and receive calls with Telzio. Telzio works with all SIP compatible apps.

Search...

Getting Started with Telzio (/support/getting-started)

Account
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Calling
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- Call Destinations (/support/call-destinations)
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- **Outbound Calling (/support/making-outbound-calls)**
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Developers (https://developer.telzio.com)

(https://telzio.com)

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Exhibit H

Home (/) Products (/products) Business Phone Systems (/products/business-phone-systems)

On-Site Business Phone Systems (/products/business-phone-systems/on-site)
Other (/products/business-phone-systems/on-site/other)

Other (/products/business-phone-systems/on-site/other)

Mitel SIP Overview (/products/business-phone-systems/on-site/other/mitel-sip-overview)

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Mitel Connect Small Business Edition 100 (/products/business-phone-systems/on-site/other/mitel-connect-small-business-edition-100)

| Mitel SIP Overview (/products/business-phone-systems/on-site/other/mitel-sip-overview)

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MiVoice Wireless Messaging Gateway WSM3 (/products/business-phone-systems/on-site/other/mivoice-wireless-messaging-gateway-wsm3)

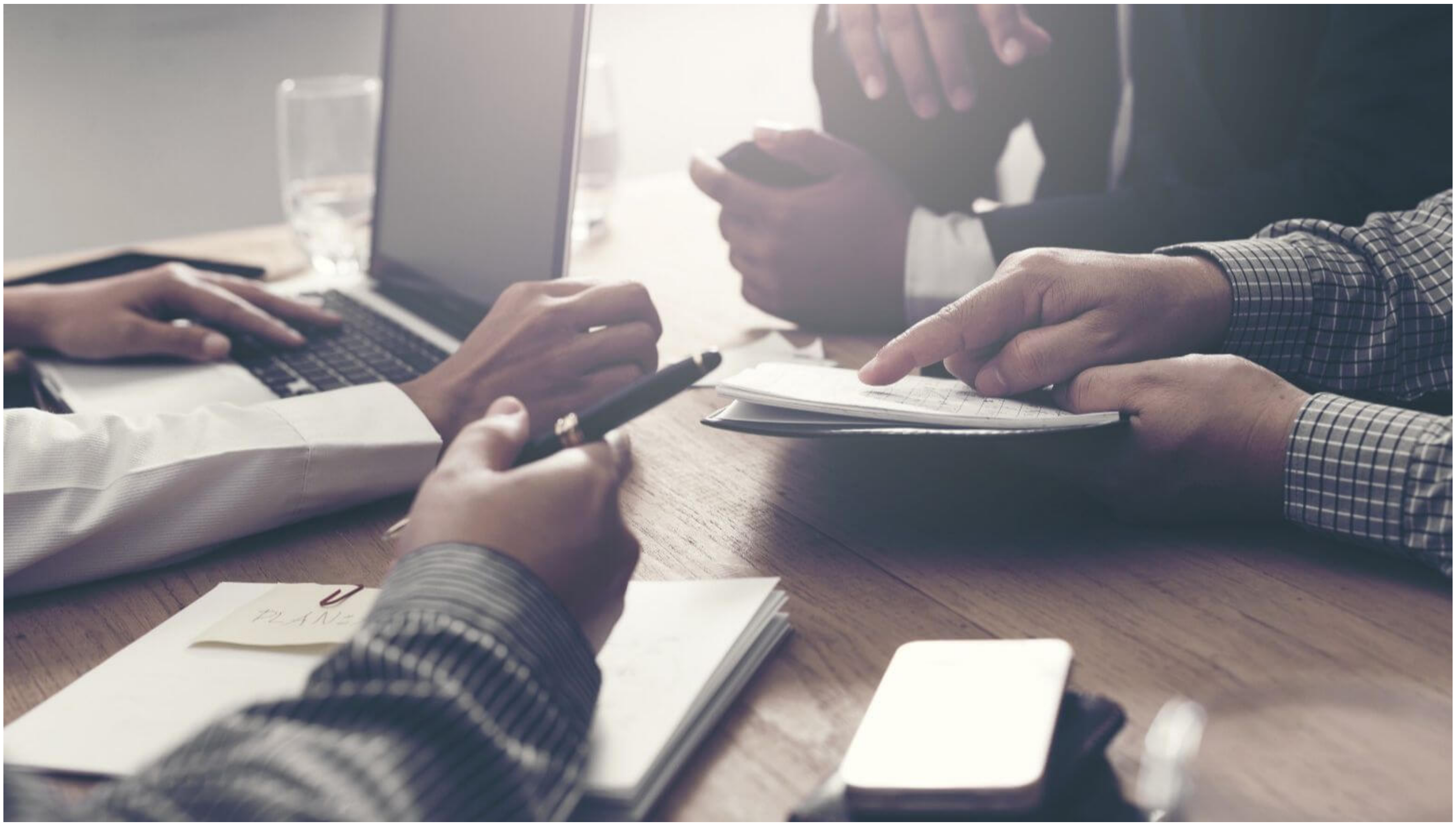
Strata CIX (/products/business-phone-systems/on-site/other/strata-cix)

Voice Switches (/products/business-phone-systems/on-site/other/voice-switches)

MITEL SIP OVERVIEW

MITEL SIP OVERVIEW

Cut costs. Keep your current phone system.



Whether you're looking to begin the transition to the cloud or simply want to save on telecom, Mitel SIP delivers high quality, reliable digital voice service without the costs of hardware.

KEY BENEFITS

CLOUD BENEFITS

Mitel SIP trunking services allow you to take advantage of key cloud benefits while keeping your current on-site, cloud or hybrid PBX phone system, making it easy to cut costs, improve business continuity and support business growth.

PERFECT ENABLERS OF UC STRATEGIES

Combine all your communications requirements into a single manageable strategy.

SPEED OF DEPLOYMENT

With our rapid turn-up process and knowledgeable support teams, getting started with SIP trunking is faster and simpler than ever.

TECHNOLOGY

SIP trunks take advantage of the latest technology, ensuring you can easily adapt as required.

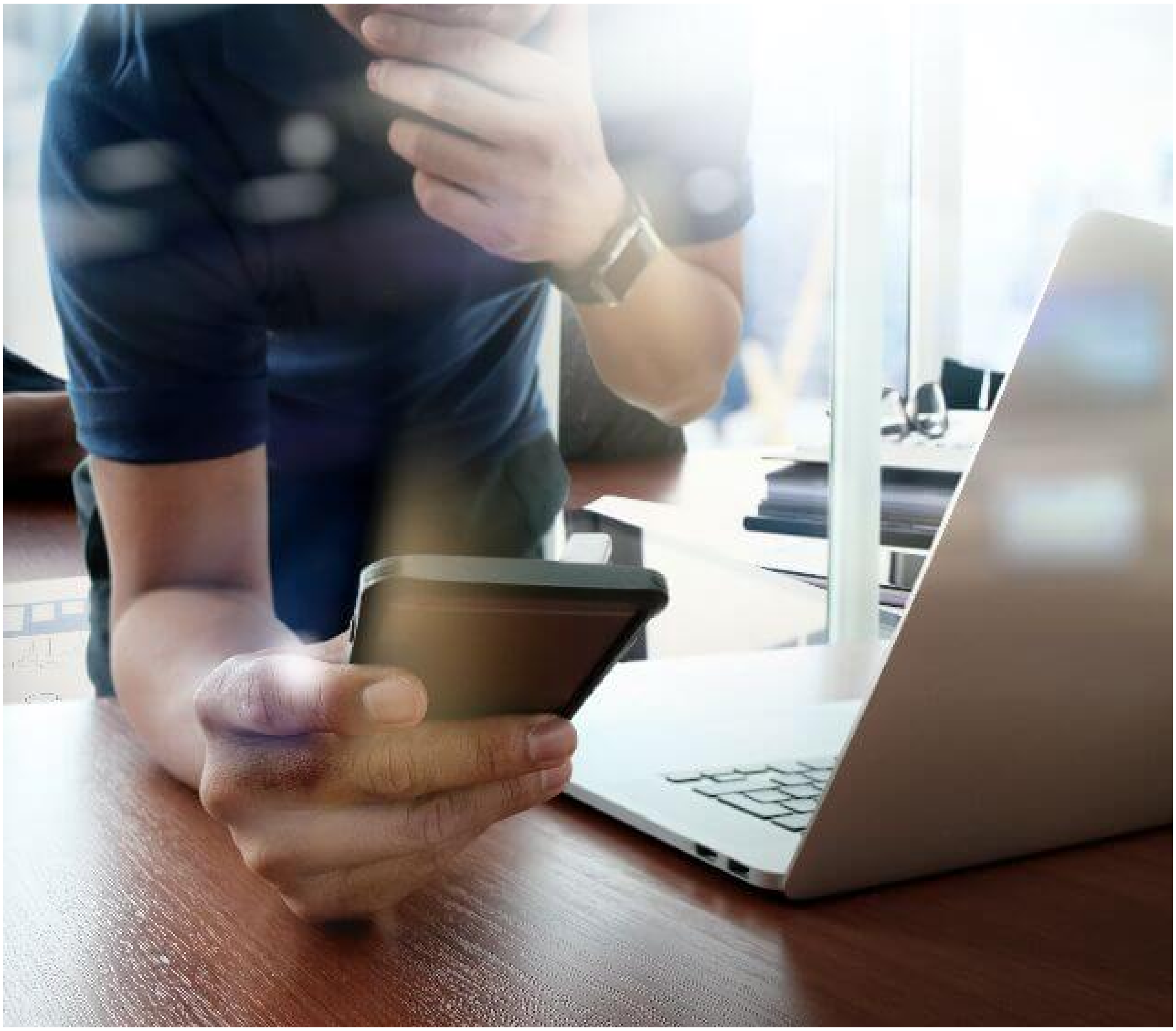
IS THIS PRODUCT RIGHT FOR YOU?

CONTACT SALES ▸ (/CONTACT)

CARRIER-CLASS NETWORK

Backed by our carrier-class cloud network, Mitel SIP delivers high-quality digital voice service you can rely on. Our large global footprint provides extensive coverage and clear call quality, making it ideal for multi-national businesses.

- Global Footprint – Ensure a consistent experience for global users and customers with ITFS coverage in 113 countries and DIDs in 59 countries.
- Scale On-Demand — With Mitel SIP, your technology can grow with your business. Call capacity can be increased on-demand to support unlimited concurrent calls and fluctuations in call volumes, while ensuring you never pay for more than you need.
- Uptime – Built on our carrier-class global network and backed by enterprise-level Quality of Service (QoS) SLAs, Mitel SIP helps companies of all sizes improve business continuity.



IMMEDIATE TELECOM SAVINGS

Lower telecom costs while maximizing the investments you've made in your existing IP phone system with SIP trunking.

- Competitive standard and toll-free per minute rates
- No long-term contracts — usage-based and bundled pricing options
- Eliminate common monthly T1 charges for instant savings
- Lower costs by eliminating the need for added telecom infrastructure and maintenance

SIP TRUNKING: THE DIGITAL VOICE ADVANTAGE

The best of the cloud with your current VoIP phone system.

- Better Business Continuity — Run your business worry-free. Our carrier-class network includes multiple levels of redundancy to ensure greater service reliability.

- High Voice Quality — Mitel SIP supports G.711, G.729 and T.38 codecs to deliver the highest possible voice quality for clear, crisp calls.

- Single Source Provider — Mitel makes it easy to simplify operations and consolidate vendors by offering UC, contact center and SIP trunking.

INTERESTED IN PURCHASING THIS FOR YOUR BUSINESS?

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800-722-1301 (tel:800-722-1301)

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