

BALLER STOKES & LIDE

A PROFESSIONAL CORPORATION

2014 P STREET, N.W.

SUITE 200

WASHINGTON, DC 20036

(202) 833-5300

FAX: (202) 833-1180

June 19, 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:

On June 18, 2019, Brannon Buck and Christopher Driver of Badham & Buck; Roger Schneider of Expert Discovery; and Jim Baller and Sean of Baller Stokes & Lide, all representing the 911 districts for Autauga County, Calhoun County, Mobile County, and the City of Birmingham in Alabama (“Districts”) participated in separate meetings with the following members of the Commission staff:

- Nirali Patel and Brett Baker (an intern) of Chairman Pai’s Office.
- Travis Litman of Commissioner Rosenworcel’s Office.
- Randy Clarke of Commissioner Starks’ Office.
- Michele Berlove, Michael Ray (via conference call), John Evanoff (via conference call) of the Wireline Competition Bureau; and Elizabeth Cuttner and Sean Saper (an intern) of the Public Safety and Homeland Security Bureau.

The purpose of the meetings was to discuss the Districts’ positions on the pending petitions for declaratory ruling in the above captioned proceeding. The Districts’ remarks were consistent with its petition and its filed comments in the proceeding.

A copy of materials presented at the meeting demonstrating that telephone carriers routinely market IVoIP enterprise solutions with burstable and dynamic sharing capabilities allowing for large volumes of simultaneous calls is attached.

If you have any questions or need additional information, please do not hesitate to contact me.

Sincerely,

Sean A. Stokes

Sean A. Stokes

cc: Nirali Patel
Travis Litman
Randy Clarke
Michele Berlove
Michael Ray
John Evanoff
Elizabeth Cuttner

**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 XXX 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.**

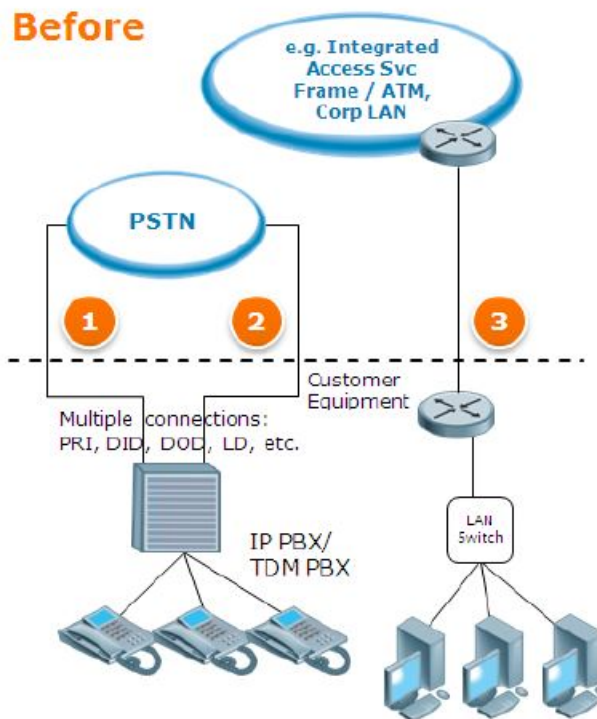
When emergency services are called over a SIP trunk, delivery of the call to the correct selective router must be verified with the provider. Unlike PRI circuits that terminate at the local LEC, SIP trunks might not have a physical connection with the local LEC and as a result will not automatically 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.

AT&T IP Flexible Reach Service

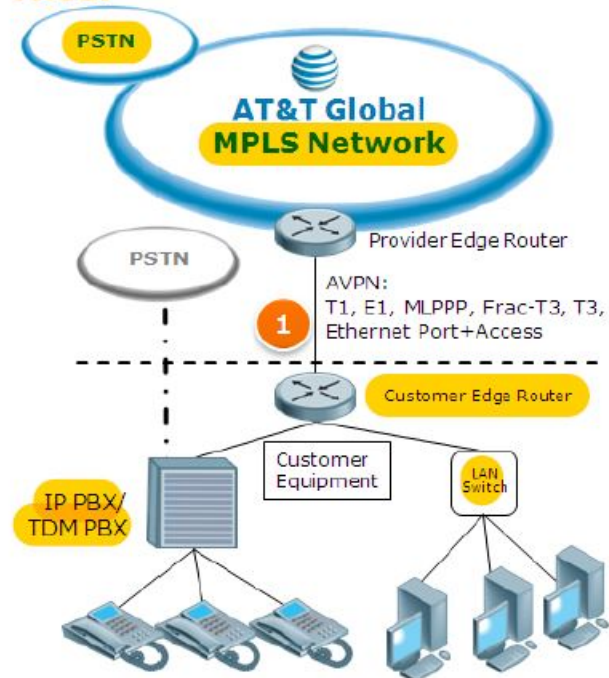
- I. Service Overview
- II. Service Components, standard and options

Voice and Data **Convergence** with IP Flexible Reach

Before



After



I. Service Overview

AT&T's Business Voice over IP ("AT&T BVoIP") portfolio of services enable the transmission of voice telephone calls in IP format over a BVoIP compatible transport service to and from Sites where both AT&T BVoIP and a BVoIP compatible transport service have been installed.



8 What is AT&T Business VoIP (BVoIP) with CoS?

The AT&T BVoIP service previously offered a VoIP and Data service where unused VoIP bandwidth would be dynamically allocate to data traffic in a lower priority data class. With COS enhancement, AT&T BVoIP customer can now select any of 17 CoS profiles with up to four classes. Multiple classes give BVoIP customers additional control for traffic separation of their data traffic. In addition, all applications including VoIP can be assured to have sufficient bandwidth by selecting a profile with the appropriate bandwidth allocation.

The purpose of this section is to capture the impact to the CPE router needed to support BVoIP with CoS enhancement and to provide new guidelines on the maximum concurrent calls for each CoS profile.

8.1 Guidelines and Number of Concurrent VoIP Calls

Using the port speed and the bandwidth percentage allocation for the real-time class, the number of concurrent VoIP calls can be calculated. BVoIP customer are given the option to select the CoS profiles with real-time allocation (#102 - #117). However, IOS Firewalls are not supported with these additional supported CoS profiles. The tables below capture the number of concurrent voice

8.1.1 TDM PBX with PRI

The following table captures the maximum number of concurrent voice calls for a TDM PBX using G729 codec and a Primary Rate Interface (PRI).

Profile #	Profile Name	T1	FT3 10M	FT3 15M	FT3 20M	FT3 25M	FT3 30M	FT3 35M	FT3 40M	T3
101	90% RT 0/0/100 Data	46	230	230	230	230	230	230	230	230
102	80% RT 80/10/10 Data	42	230	230	230	230	230	230	230	230
103	80% RT 60/30/10 Data	42	230	230	230	230	230	230	230	230
104	80% RT 40/30/30 Data	42	230	230	230	230	230	230	230	230
105	60% RT 80/10/10 Data	30	214	230	230	230	230	230	230	230
106	60% RT 60/30/10 Data	30	214	230	230	230	230	230	230	230
107	60% RT 40/30/30 Data	30	214	230	230	230	230	230	230	230
108	50% RT 0/0/100 Data	24	180	230	230	230	230	230	230	230
109	40% RT 80/10/10 Data	18	142	230	230	230	230	230	230	230
110	40% RT 60/30/10 Data	18	142	230	230	230	230	230	230	230
111	40% RT 40/30/30 Data	18	142	230	230	230	230	230	230	230
112	20% RT 80/10/10 Data	10	70	100	130	160	200	230	230	230

IP Trunking Services With Burstable Enterprise Shared Trunks (BEST)

The Next Generation Trunking

A New and Better Way to Manage Trunking Resources

Use idle trunk capacity in one location to accommodate an increase in traffic from another location

Purchase fewer concurrent calls at each location and share resources to provide time of day benefits and peak usage management

Leverage “locationless” concept of trunk utilization, not feasible with standard TDM lines with physically defined connections

Combined with:

- Single vendor solution
- Nationwide standard local services offer
- Flexible IP-based failover and load sharing options

Which means a **comprehensive solution** for all your enterprise trunking requirements



Burstable Enterprise Shared Trunks (BEST)

Meeting Your Business Needs

BEST is a new service enhancement for Verizon VoIP services which allows you to leverage trunking resources.

Verizon dynamically monitors use of concurrent call ports across all your locations and allows a location to 'burst' over the quantity ordered if ports are idle at other locations

You can better utilize your voice networks by accessing pooled resources across your enterprise



IP Trunking Services with Burstable Enterprise Shared Trunks (BEST)

The Next Generation Trunking

A New and Better Way to Manage Trunking Resources

Use idle trunk capacity in one location to accommodate an increase in traffic from another location

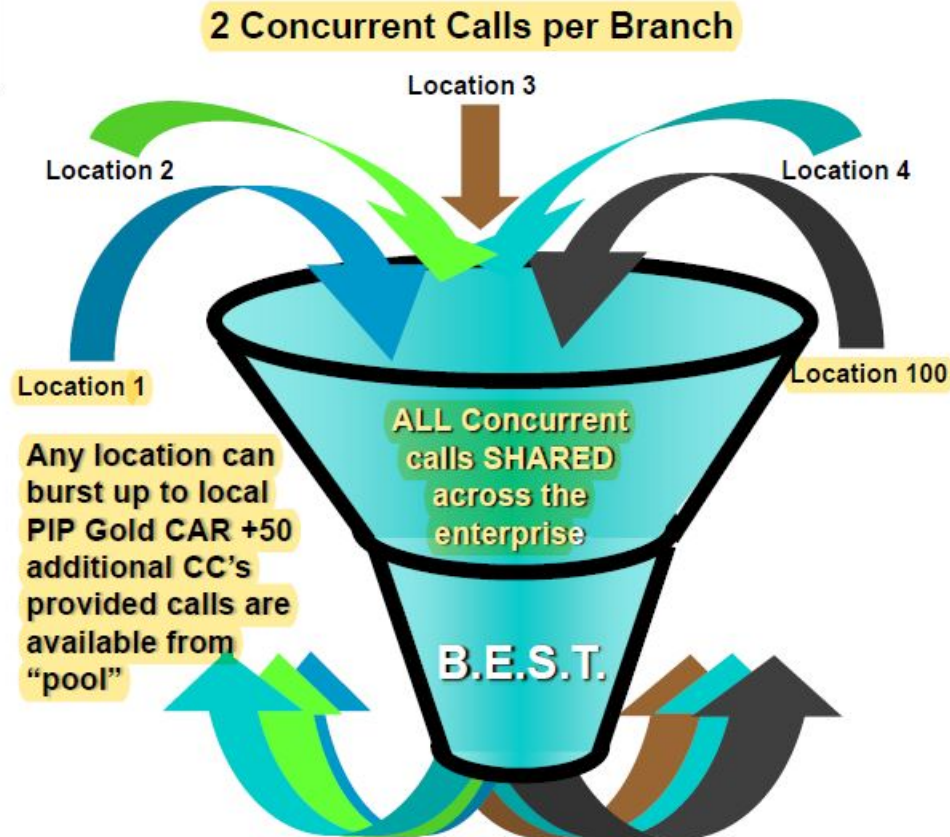
Purchase fewer concurrent calls at each location and share resources to provide time of day benefits and peak usage management

Leverage "location-less" concept of trunk utilization, not feasible with standard TDM lines with physically defined connections; i.e. better use of assets

Reduces hard assets required to support locations

Combined with:

- End to End Single vendor solution
- Flat Rate Long Distance
- Flexible IP-based failover and load sharing options
- Flexible Rate Plans (Local+LD or LD)



IP Trunking with BEST: Savings Example

This enterprise with 1100 locations has realized savings of **\$219,170** or **30%** in monthly costs

TDM Trunking

13,200 Trunks



12 Lines
per
Location



IP Trunking with BEST

**5,500 Shared Concurrent
Call Pool**

5 Lines
per
Location



Monthly Cost:

\$50/line x 1100 stores = \$660,000

Monthly LD (1700 min avg) = \$55,000

TOTAL Current monthly \$715,000

Monthly Cost:

5,500 Concurrent Calls @\$32 = \$176,000

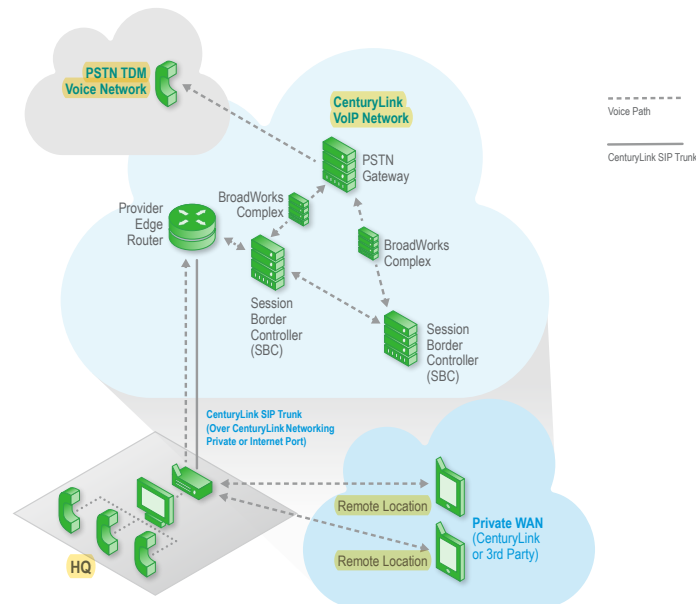
Monthly LD (1700 min avg) \$28,055

Additional PIP circuit cost \$291,775

TOTAL new monthly \$495,830

CenturyLink IQ® SIP Trunk: Centralized Deployment

Simplify the management of your voice and data traffic by consolidating PRIs and POTs with our centralized IQ SIP Trunk solution. CenturyLink IQ SIP Trunk enables you to make and receive local voice, dedicated long-distance, domestic and international toll-free service, and Enhanced V911 calls across a single broadband connection.



Extend the capabilities of your existing, legacy PBX hardware while converging your separate voice and data networks into a single, robust IP data and voice network. IQ SIP Trunk works in conjunction with CenturyLink® MPLS Networking, providing a private, fully interoperable and scalable suite of wide area network (WAN) services. IQ SIP Trunk delivers increased functionality, improved call quality, better security, and simplified network management and is available nationwide. CenturyLink backs its IQ SIP Trunk service with a 100% voice availability SLA. Leverage simple end-user and administrative management portal, and robust quality of service (QoS) to help ensure that time-sensitive network traffic is prioritized and routed correctly.

Technical Specifications

- Consolidates voice and data traffic over a single, robust OC-192 IP data backbone
- Incorporates the CenturyLink IQ Networking managed suite of wide area network (WAN) services
- Advanced IP-centric, multi-protocol label switching (MPLS)
- Can be provided over third-party, IP-enabled networks to sites without IQ MPLS connectivity
- Includes PSTN/PRI Failover & Administrator/End-user Portals

Centralized deployment

- Provide a centralized trunking service at HQ by consolidating PRIs and POTs
- Branch offices reach service through the main site via their private, QoS-enabled WAN
- E911 or V911 services are still provided with local service
- Consolidated of voice and data at a centralized location
- Ease of management
- Reduction of PSTN access