Ingate Systems, a pioneer in SIP trunking, now brings new important technology and products to WebRTC.

click to call

1-8XX-REPLACE

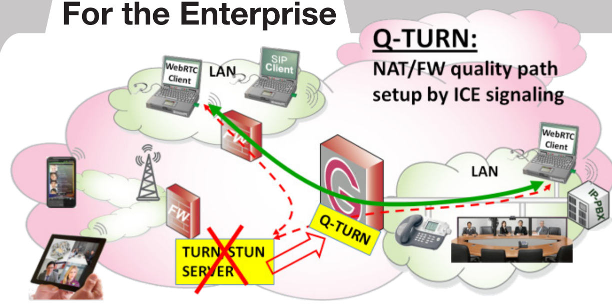
For Carriers, Service Providers and Enterprises:

$$$ The 15 Million Dollar Box $$$

Presenting:

Q-TURN for the Enterprise
TURN server in the Ingate SIParator® E-SBC to enable and give quality to WebRTC and SIP-based real-time communication.

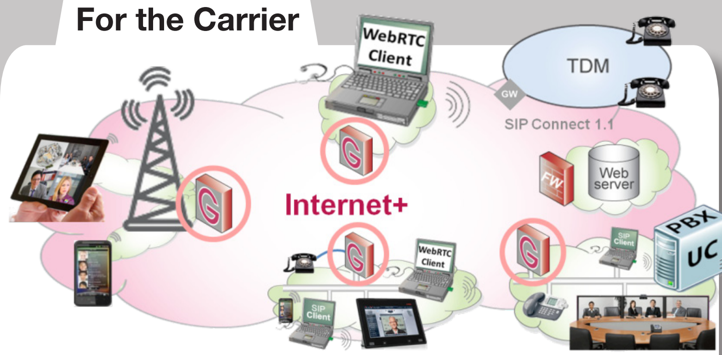
For the Enterprise



Reliable WebRTC Access
(Even With Restrictive Firewall)

- Priority, Traffic Shaping for QoS
- Diffserv and RSVP Over the Network
- Acces Control - Authentication

For the Carrier



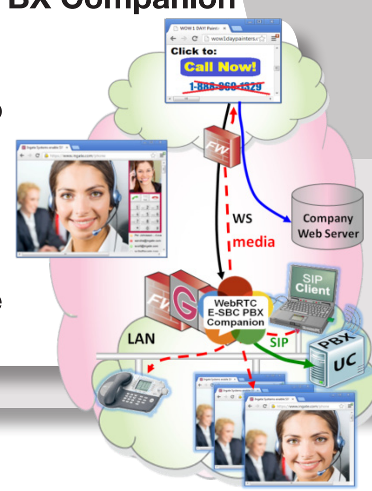
Reliable Quality WebRTC and Accounting

- Authentication and RTC Access Control
- QoS: Priority, Traffic Shaping and Network
- Valuable RTC Traffic Counted Separately
- Mobile, Enterprise, SMB and Residential

Q-TURN for Carriers and Network Providers
Offer a "WebRTC-Ready" broadband access!

WebRTC and SIP PBX Companion

- Click to Dial-In
- Pass a link to call into a meeting
- Browser as PBX Soft Client
- Home workers and road warriors just use the browser



WebRTC and SIP PBX Companion
Give the enterprise PBX and UC solution all the benefits of WebRTC. The PBX, call center and UC solution infrastructure will be integrated with the WebRTC world.

...and Supporting Carrier's Great Interest in Providing WebRTC Services as a Continuation of Voice Services

Ingate's mission is to enable the best access for telephony, global real-time and unified person-to-person communication for everyone.

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WebRTC is real-time communication directly between Web browsers or rather between persons on the Web. Just click on the person or service when found on the Web, get connected with HiFi voice and HD telepresence video - everywhere.

That is a giant step from stretching out for a telephone – dial a number, see if you reach the called party - and then maybe agree on some better way than pre AM-radio quality voice to communicate. In addition, WebRTC will bring video conferencing, screen sharing and data channels to our everyday communication everywhere.

Ingate's WebRTC Related Technology and Products

Q-TURN for Firewall Traversal and Quality

Q-TURN is Ingate's novel technology for guiding the real-time traffic through a TURN server between a private network and the Internet to assure real-time traffic quality in data crowded networks. It also enables WebRTC into enterprise LANs with restrictive firewalls otherwise blocking such traffic.

Q-TURN for the Enterprise

Firewall vendors are welcome to integrate Ingate's Q-TURN technology on an OEM basis. Q-TURN is also part of the Ingate WebRTC & SIP PBX Companion product to assure quality communication between WebRTC browsers on the LAN and WebRTC browsers on the Internet.

Carriers Can Deliver Q-TURN, UC and WebRTC Communication as a Continuation of SIP Trunks

Carriers should have Q-TURN in every access router providing a protected private Internet or OTT access (to the LAN). Q-TURN enables quality-assured, measurable WebRTC real-time communication separate from the data traffic.

Carriers can provide click-to-call links and buttons, replacing toll free 800 numbers, for the enterprise website and superior video conferencing clients for local and remote UC and PBX usage.

Access router CPE vendors are encouraged to incorporate Ingate's Q-TURN technology. Q-TURN should also be used in the mobile DPI firewall/router behind the cell tower for quality and accounting of WebRTC traffic over the OTT channel.

WebRTC & SIP PBX Companion for PBX and UC Solution Vendors

The Ingate WebRTC & SIP PBX Companion brings all features of WebRTC to the enterprise SIP PBX or UC (Unified Communications) solution. The Web browser becomes a high-end video SIP client, available everywhere you can surf. You can also pass a link to be called and place click-to-call buttons on the enterprise Web page, connecting to the SIP/UC PBX or call center infrastructure, saving huge amounts by replacing toll free 800 numbers. (Without the Companion, WebRTC bypasses the enterprise SIP PBX/UC infrastructure.)

WebRTC & SIP Companion for IMS/RCS/Joyn Services

The same product as the PBX Companion, placed at the customer site, can also be the WebRTC gateway to the service provider's multimedia real-time communication service. The service provider simply writes a Web app for its service.

Compared to centralized WebRTC-IMS gateways, there are no scaling problems, no enterprise NAT/firewall traversal problem and quality is assured even with heavy data traffic.

Simultaneously, the same Companion CPE can SIP trunk PBXs, provide general SIP to the LAN and be the PBX Companion.

WebRTC Gateway for OTT Telephony Service Providers

The quality, superior connectivity and NAT/firewall traversal of the Ingate WebRTC–SIP gateway, makes reliable, volume, multimedia OTT telephony services possible. Placed centrally, running on powerful x86 servers, multiple Ingate WebRTC gateways scale to any size and only need to gateway the media when interoperating with the POTS world.

For Next-Generation Volume-Deployed CPE Access Routers: Cable, DSL, Ethernet, Fiber and 4G Routers

Ingate's Q-TURN, E-SBC and the full WebRTC & SIP PBX Companion can be integrated into embedded devices such as the access router/modem itself, for carrier mass-volume deployments. Ingate's expertise in highly advanced, reliable, secure and efficient real-time communication software for embedded systems is exemplified by Ingate's own IX78 E-SBC for SIP trunking of up to 50 concurrent calls, containing also a full enterprise firewall/router, VPN, IP TV and more, running on an embedded ADSL2+/Ethernet modem chipset.

Session-based license models for SMB to residential. Quality assured (QoS) real-time communication and CDRs with quality metrics.