Far South Networks

WebRTC enabled CPE appliance

Com.X provides a high performance, “3-in-One” package for managing voice, data and messaging services for the SMME user

Com.X integrates Analog & IP voice and provides enterprise class data services over IP WAN, e.g. ADSL2+, 3G/LTE, CDMA & WiFi

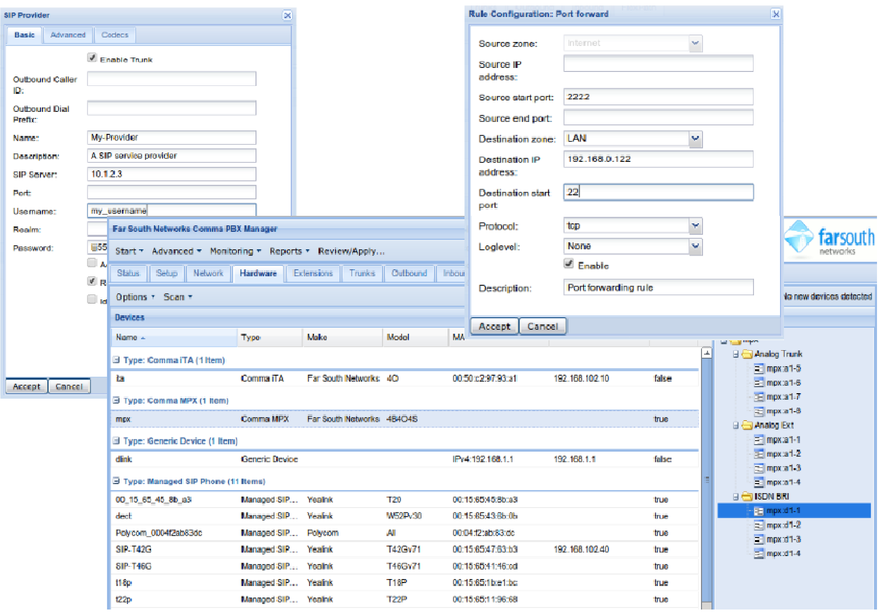
Voice and Data Convergence:

“3-in-One” package: Router/Firewall + VoIP Gateway + PBX

- Security: Integrated firewall & router including VPN support, DOS resilience
- Connectivity: Dynamic, multilevel call routing over all available trunks. Highest quality calls maintained by dedicated hardware echo cancellation and best of breed QoS algorithms
- Services: Managed Voice and Data Services
  - Provisioning: TR-069
  - Configuration: Template generation, backup and restore
  - Monitoring and reporting: Nagios integration
  - Remote management and licensed s/w upgrade
  - PBX services failover: Configuration interop with Cloud/ IP Centrex platforms
  - Roadmap to WebRTC convergence

Powerful, flexible and easy to use:

Com.X integrates the open-source applications Ubuntu Linux, Asterisk and FreePBX, within a rich, wizard-like web GUI using Java and Google Web Toolkit. Additional major features include SIP phone auto-provisioning, custom dial-plan routing (FlexPath) and Integrated Access Device configuration – networking, firewall, PPPoE, VLAN, VPN and more.



- **Router / Firewall**
- **VoIP Gateway**
  - Analog, ISDN, SIP, WebRTC
  - Back-to-Back User Agent (B2BUA)
  - QoS: TOS, Traffic & Services Classes
- **WebRTC Gateway**
  - Real time streaming between end-points
  - Opus CODEC (voice)
  - VP8 CODEC (video) – roadmap
- **Full PBX feature set**
  - IVR, Conferencing, Queues, Voice Mail-to- email
  - Call Recording
  - Analog & IP phone support with auto-provisioning
- **TR-069:** Remote configuration, management and monitoring
- **Seamless expansion:** Comma iTA telephony port adapter
- **Certification:** TBR3, TBR4, EN60950, CISPR22 Class B

Com.X5 brief

Users: 50 to 125+ extensions
VoIP trunks: 8 or 16 * G.729
ISDN: 4 or 8 Basic-rate ISDN
Analog FXS/FXO: 2 to 24
LAN/WAN: 4 GbE
Storage: 16GB SD card or 64GB SSD
Optional: ADSL2+ WiFi, 3G/LTE, X.21 sync serial
Roadmap to: DECT, 6 handsets, 4 channels

Com.X10 brief

Users: 125 to 250+ extensions
VoIP trunks: 30 or 60 * G.729
ISDN: 1 or 2 Primary-Rate, 4 or 8 Basic-rate ISDN
Analog FXS/FXO: 8 or 16
LAN/WAN: 3 GbE
Storage: 64GB SSD
Optional: WiFi, 3G/LTE, X.21 sync serial

Comma iTA brief

LAN based PSTN adapter for Com.X
Primary-Rate ISDN: 1 to 4 ports
Basic-rate ISDN: 4 to 8 ports
Analog FXS/FXO: 8, 16, 24 or 32 ports
LAN 1 10/100Base-Tx

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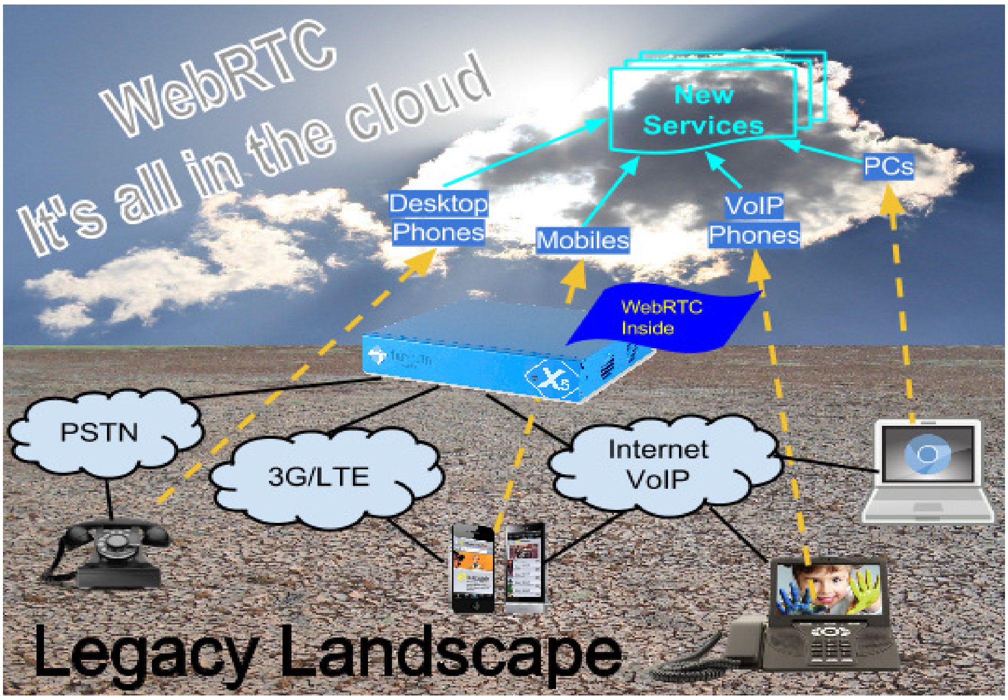
WebRTC enabled IP PBX: What does this mean to me?

Imagine a world where your business and personal phone, computers and smart devices can all communicate on a common platform. Imagine it is easy to add real-time voice, video and peer-to-peer data sharing to your web application. That's the vision of WebRTC.



A WebRTC enabled PBX or Gateway bridges the divide between the current generation of PSTN, mobile and VoIP devices and services, and the emerging Real-Time World Wide Web.

WebRTC provides a universal BYOD environment at work, at home, anywhere in the world. Gone is the need for the business IT Administrator to install, maintain and secure disparate device applications.

Emerging real-time enabled cloud applications will facilitate highly flexible associations between voice and video capable entities (people, companies and services) and connect them to businesses and social communities, in a manner similar to today's social networks. The addition of WebRTC capability and convergence with legacy services means that real-time presence management and true "anywhere, anytime" reachability are finally possible.

For example, Bob may be identified by a phone number, an e-mail address and a Google account. Bob is currently out of town, and the Web registry is aware that Bob is reachable via his mobile phone, WebRTC client on his laptop and e-mail. Sue needs to call Bob, and is authorized to call Bob because of an agreed upon community association (she works for the same company). She does so by entering his e-mail address into a Web App and selecting Voice Call. Bob is contacted by the current best possible path, and the call is completed. Similarly, Sue could have called Bob's phone number from an ordinary phone, and Bob could have taken the call on his laptop.

The Real-Time WWW shifts paradigms:

- SIP or PSTN carriers become themselves community members in the "WebRTC Cloud", offering services to other community members (e.g. gateway service, recording service
- Devices connect Peer-to-Peer over the best path available
- Your different contact endpoints are rolled in one URI that represents YOU
- The main function of a communications service provider is now to organize and broker associations between community members
- Virtualize the phone into a WebApp or Browser

Legacy - Telco landscape	**WebRTC – in the Cloud**
Trunking	Peering
Parallelogram	Triangle / Line
Phone number	URL
Service provider	Community
Centralised	Distributed
Phone	Browser / WebApp