Established 1961

2013 WBA Broadcasters Clinic

Madison, WI

MIGRATING RADIO CALL-IN TALK SHOWS TO WIDEBAND AUDIO

Radio is the original Social Network

- Serves local or national audience
- Allows real-time commentary from the masses
- The telephone becomes the medium
- Telephone technical factors have limited the appeal of the radio “Social Network”

Telephones have changed over the years







But Telephone Sound has not changed (and has gotten worse)
This is very bad for Radio

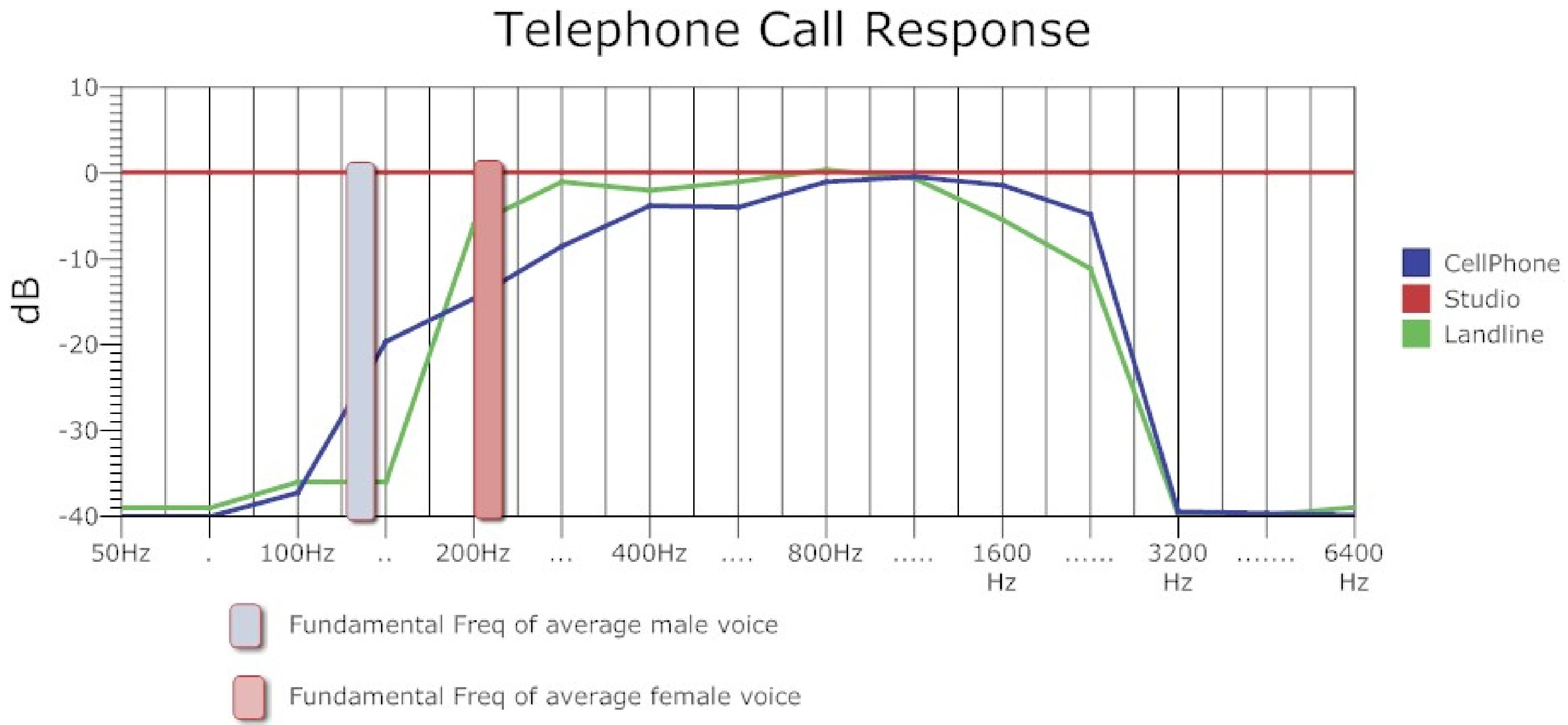
Why do phones sound bad?

- System designed for efficiency not comfort
- Sampling rate of 8kHz chosen for all calls
- 4 kHz max response
- Enough for intelligibility
- Loses depth, nuance, personality
- Listener fatigue

Why do phones sound so bad ? (cont)

- Low end of telephone calls have intentional high-pass filtering
- Meant to avoid AC power hum pickup in phone lines
- Lose 2-1/2 Octaves of speech audio on low end
- Not relevant for digital

Why Phones Sound bad (cont)



Los Angeles Times -- January 10, 2009

Verizon Communications Inc., the second-biggest U.S. telephone company, plans to do away with traditional phone lines within *seven years* as it moves to carry all calls over the Internet.

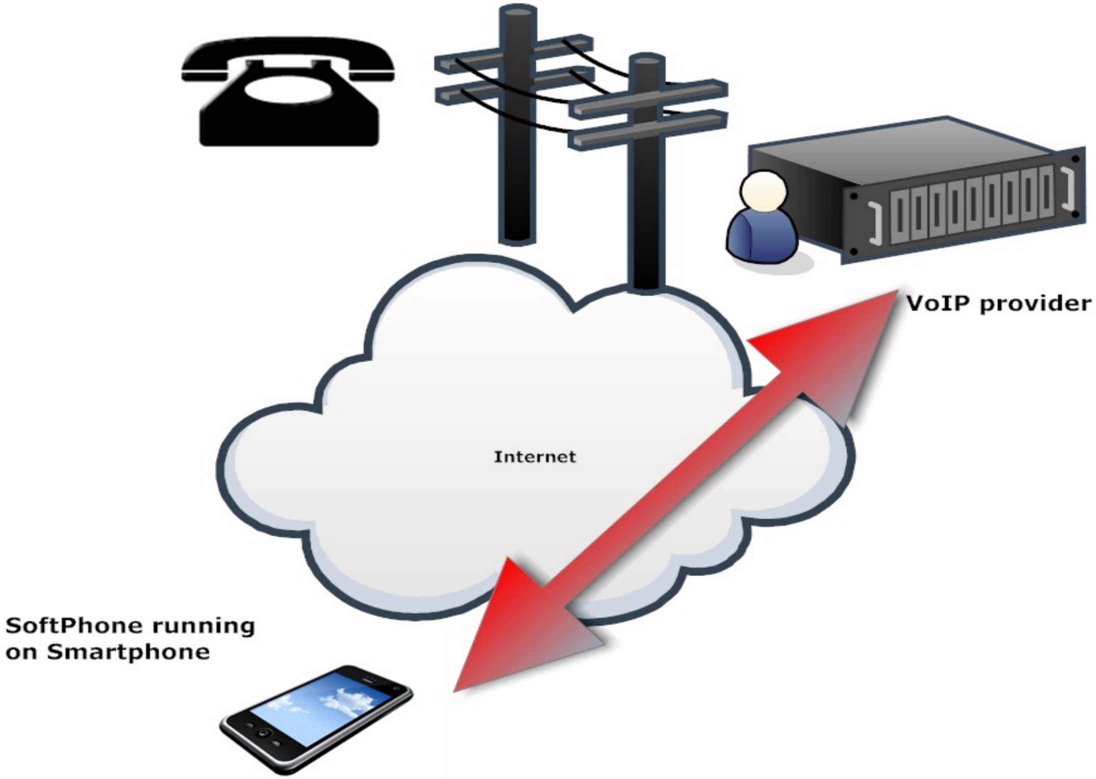
An Internet-based service can be maintained at a fraction of the cost of a phone network and helps Verizon offer a greater range of services, Stratton said.

"We've built our business over the years with circuit-switched voice being our bread and butter...but increasingly, we are in the business of selling, basically, data connectivity," Chief Marketing Officer John Stratton said.

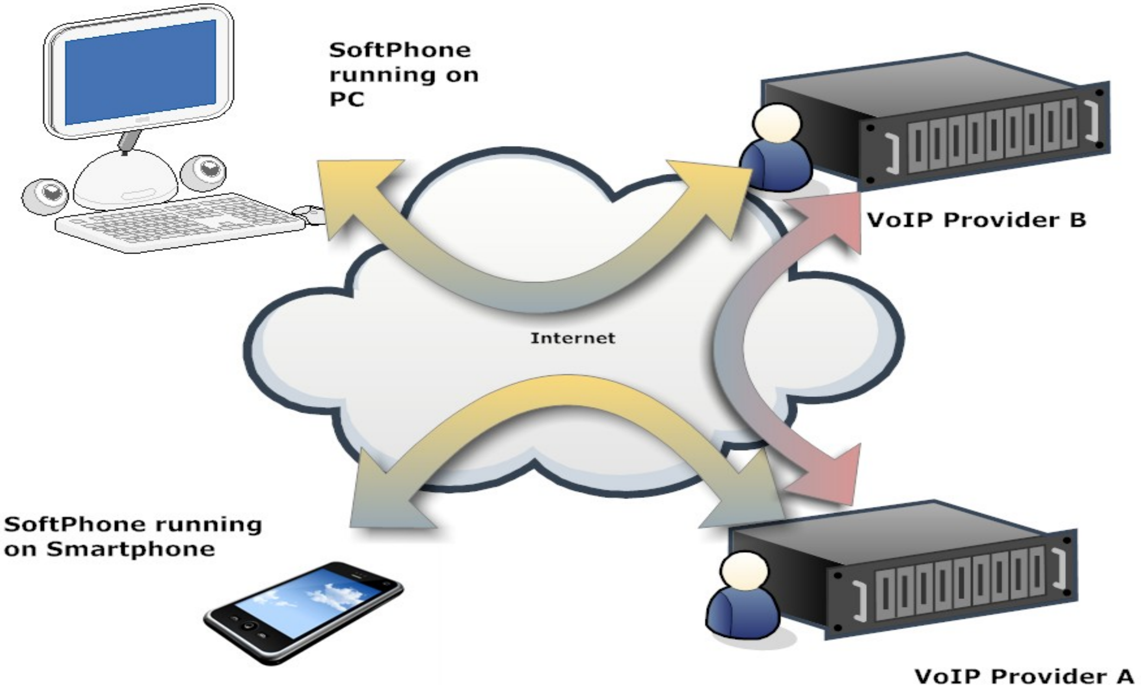
VoIP Rules the Industry

- SIP based hardware interworks
- Lower cost—easier to manage
- Hard to find non-VoIP business systems
- Delivered to homes by many providers

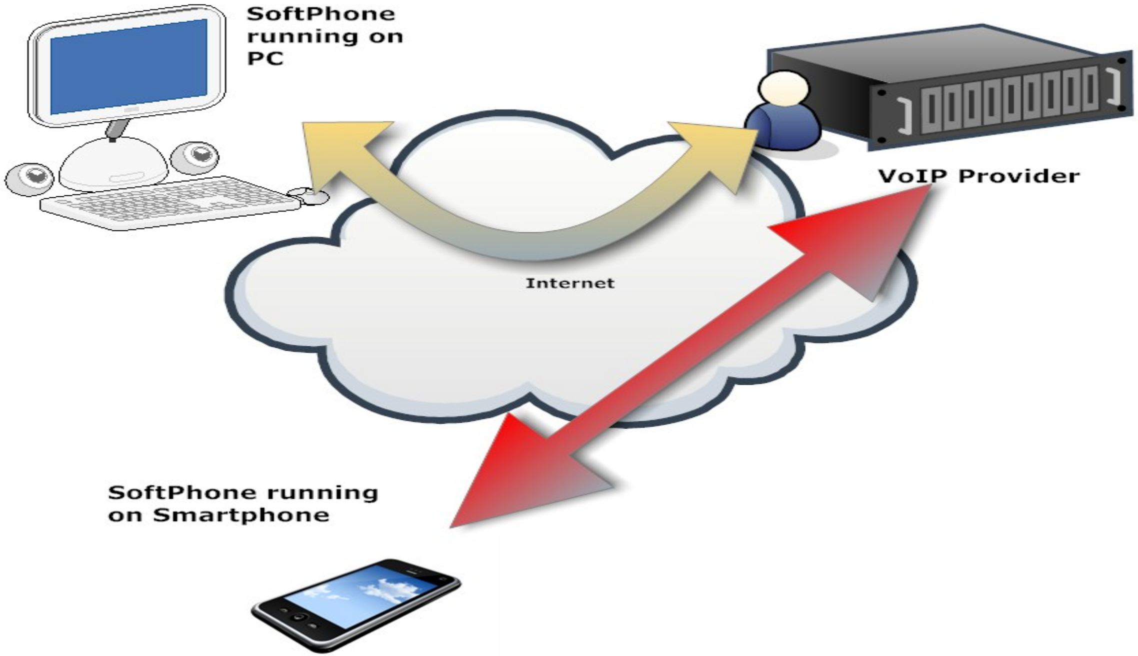
VoIP



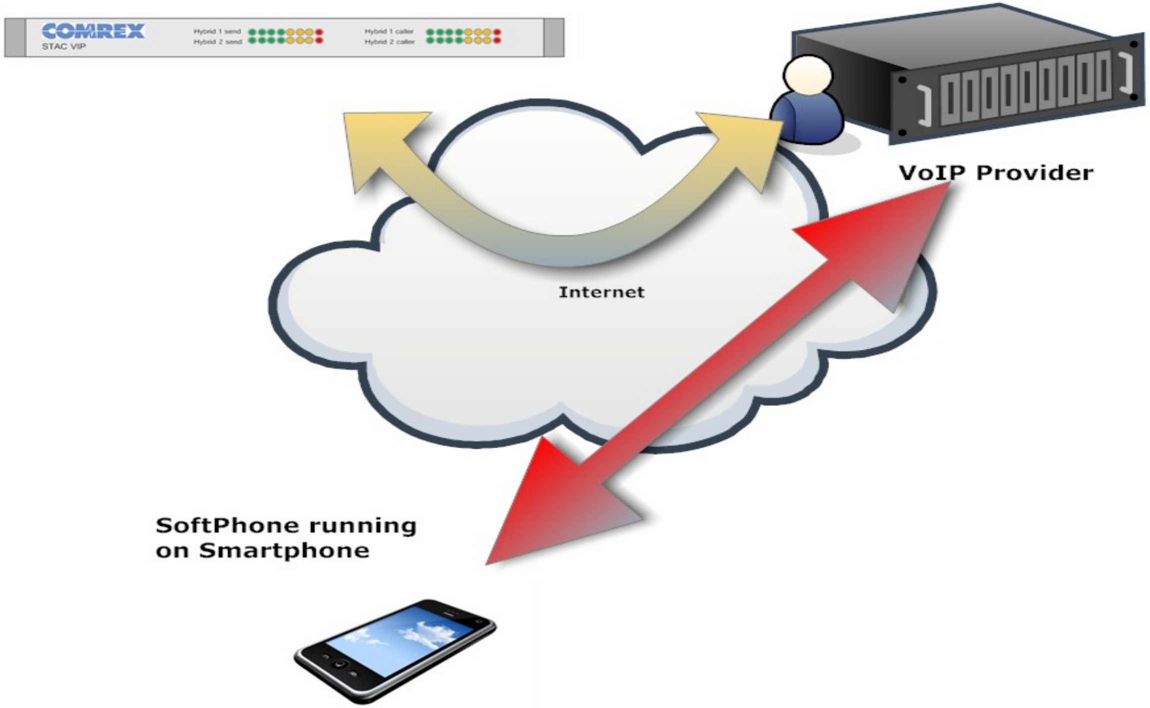
VoIP –VoIP can be made not to touch the switched telephone network



SIP calls can be made without SIP Registration



IP based talk show system

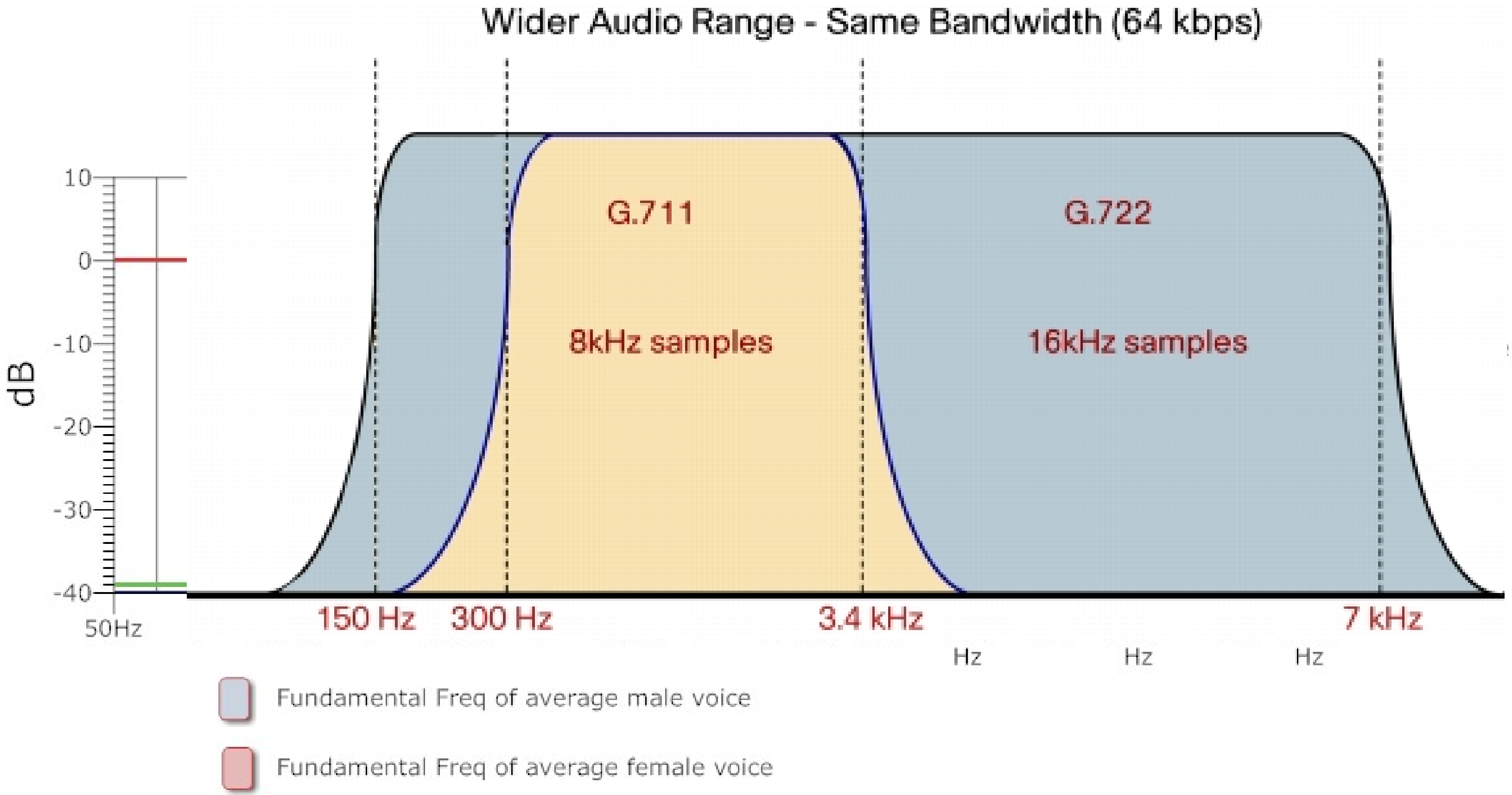


SIP URI

- Calls dialed by URI will not touch the telephone network
- Format like email username@host
- Example: allhitradio@iptel.org

Wideband Codecs for VoIP

- G.722- Most commonly supported
- G.722.1 and G.722.2- More modern, lower bitrate
- Speex- Free and Open Source alternative
- iSAC- Google owned and free, used in Gvoice
- G.711.1, G.729.1- less widely implemented


Wider Audio Range - Same Bandwidth (64 kbps)
dB
10
0
-10
-20
-30
-40
50Hz
G.711
8kHz samples
G.722
16kHz samples
150 Hz
300 Hz
3.4 kHz
7 kHz
Hz
Hz
Hz
Fundamental Freq of average male voice
Fundamental Freq of average female voice

Device Manufacturer Support:

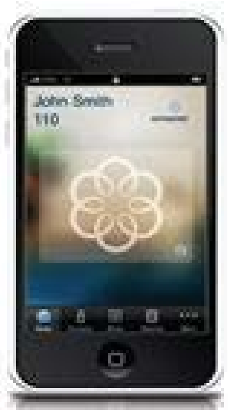
Aastra, Acme Packet, Aculab, Arris, AudioCodes, Avaya, Cisco, CounterPath, D2 Technologies, Dialogic, Digium, Ditech Networks, DSP Group, Ericsson, GENBAND, Gigaset (Formerly Siemens), Grandstream, Huawei, HTC, LG, Metaswitch Networks, Motorola, NETGEAR, Nokia, Panasonic, Polycom, Samsung, snom, Sonus Networks, Technicolor (Formerly Thompson), VTech, and WYDEVoice

Service Provider Support:

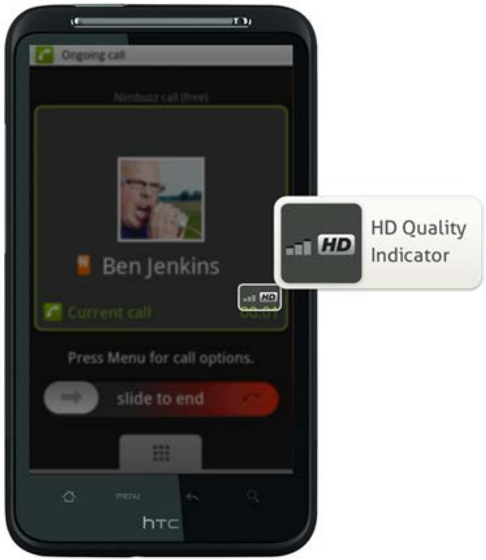
3, AT&T, BT, France Telecom/Orange, Neutral Tandem, Ooma, Portugal Telecom, SFR, Skype, Sprint, Tata DOCMO, Telekom Austria, Telecom Italia, Telstra, T-Mobile, and Verizon.

PC Softphones support G.722

“HD Voice” apps for Iphone



“HD Voice” apps for Android



What about Skype ?

- Most popular VoIP app
- Skype-Skype calls are free
- Available for all platforms—PC, Mac, Android, iPhone

Skype (cont)

- Skype-Skype calls negotiate codec
- Most calls of this type use Skype SILK wideband codec
- Limitations exist on some platforms

Bringing it together

What does a wideband talkshow system require?

Basic requirements of any talkshow system

- Ability to make and receive calls
- Ability to separate send and receive audio
- Ability to conference at least two callers on-air
- Signal processing to normalize caller audio levels and provide host domination

Basic requirements of any talkshow system (cont)

- Ability to screen calls before putting them on-air
- Ability to feed on-hold audio to callers
- Ability to provide useful information about caller intent to talent
- Simple interface allowing talent to select and drop calls

Other specific requirements

- Display and log caller ID
- Blacklist functionality
- Audio and metadata archiving
- Auto-priority functions

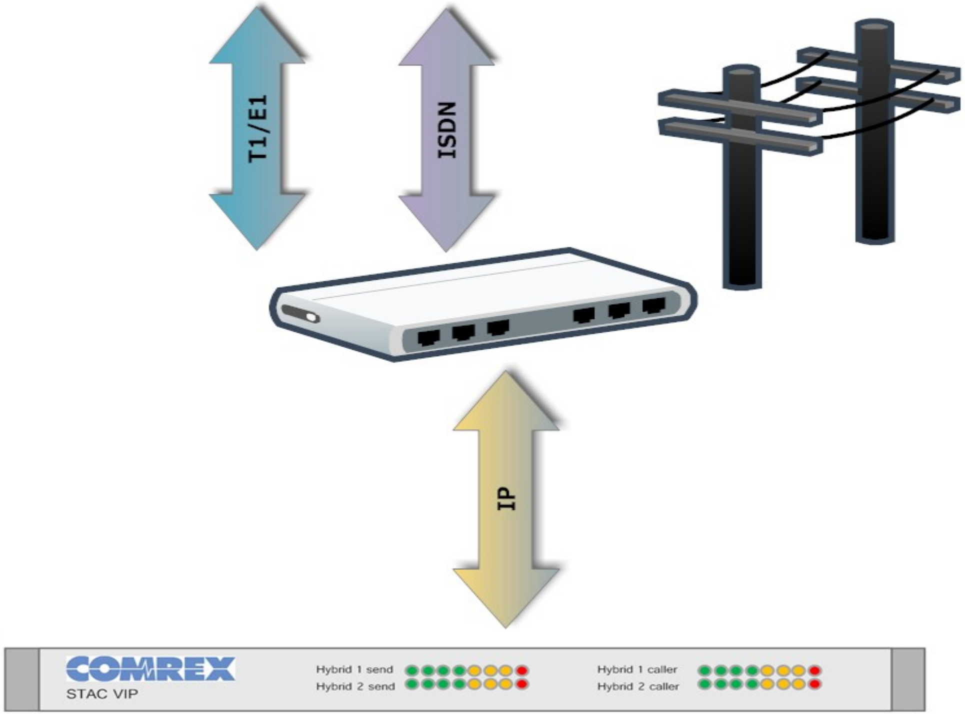
Mix of wideband and narrowband

- All these functions become more complex
- Must be equipped to handle the entire range of possible wideband options
- Handle legacy 3KHz POTS-grade calls

Legacy support

- Unlikely talk shows can migrate directly to pure VoIP
- Calls today handled by several technologies
- POTS
- T1/E1
- ISDN

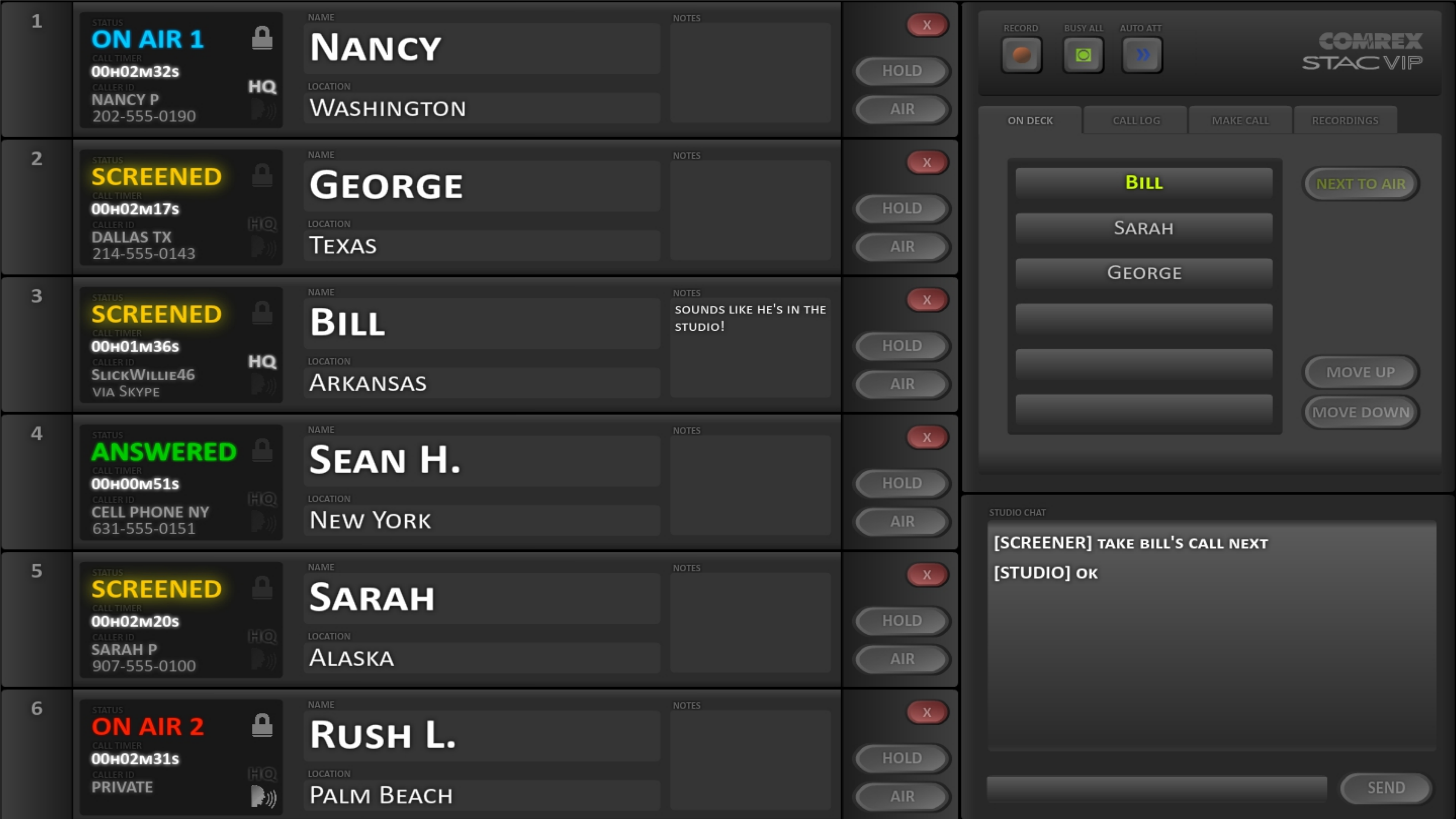
Gateways bridge legacy phone systems



POTS Gateway

The System

STACVIP


1
STATUS
ON AIR 1
CALL TIMER
00H02M32S
CALLER ID
NANCY P
202-555-0190
HQ
NAME
NANCY
LOCATION
WASHINGTON
NOTES
X
HOLD
AIR
2
STATUS
SCREENED
CALL TIMER
00H02M17S
CALLER ID
DALLAS TX
214-555-0143
HQ
NAME
GEORGE
LOCATION
TEXAS
NOTES
X
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3
STATUS
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CALL TIMER
00H01M36S
CALLER ID
SLICKWILLIE46
VIA SKYPE
HQ
NAME
BILL
LOCATION
ARKANSAS
NOTES
SOUNDS LIKE HE'S IN THE STUDIO!
X
HOLD
AIR
4
STATUS
ANSWERED
CALL TIMER
00H00M51S
CALLER ID
CELL PHONE NY
631-555-0151
HQ
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SEAN H.
LOCATION
NEW YORK
NOTES
X
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5
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CALL TIMER
00H02M20S
CALLER ID
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ALASKA
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CALLER ID
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LOCATION
PALM BEACH
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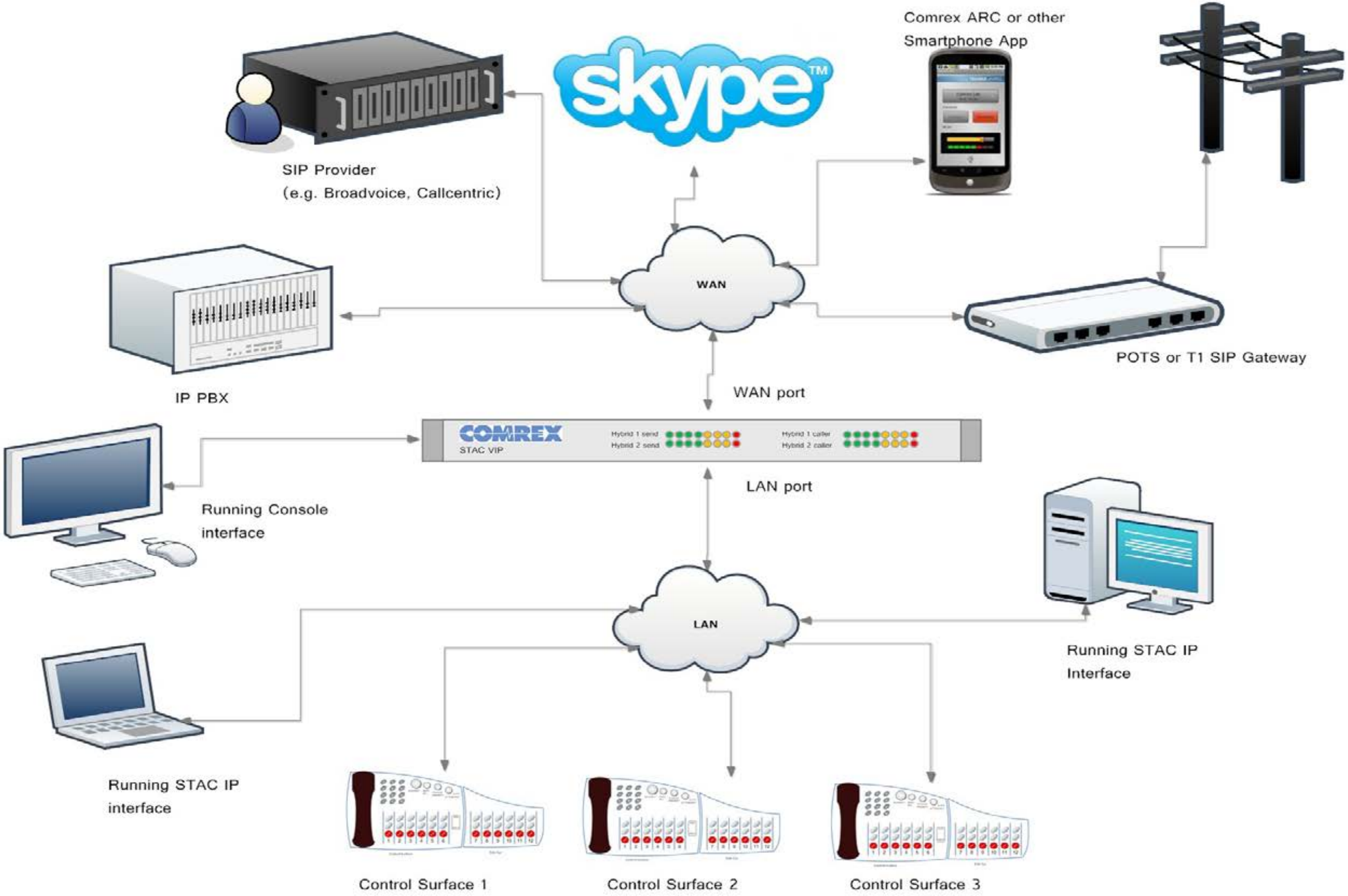
AIR
RECORD
BUSY ALL
AUTO ATT
COMREX
STAC VIP
ON DECK
CALL LOG
MAKE CALL
RECORDINGS
BILL
SARAH
GEORGE
NEXT TO AIR
MOVE UP
MOVE DOWN
STUDIO CHAT
[SCREENER] TAKE BILL'S CALL NEXT
[STUDIO] OK
SEND


Comrex ARC or other
Smartphone App
skype
SIP Provider
(e.g. Broadvoice, Callcentric)
WAN
IP PBX
POTS or T1 SIP Gateway
WAN port
COMREX
STAC VIP
Hybrid 1 send
Hybrid 2 send
Hybrid 1 caller
Hybrid 2 caller
LAN port
Running Console
interface
LAN
Running STAC IP
Interface
Running STAC IP
interface
Control Surface 1
Control Surface 2
Control Surface 3

Today’s Talk Radio

Tomorrow's Talk Radio

First Caller: G.722 softphone on Android

Second Caller: Skype on iPhone

Thank you from everyone at Comrex