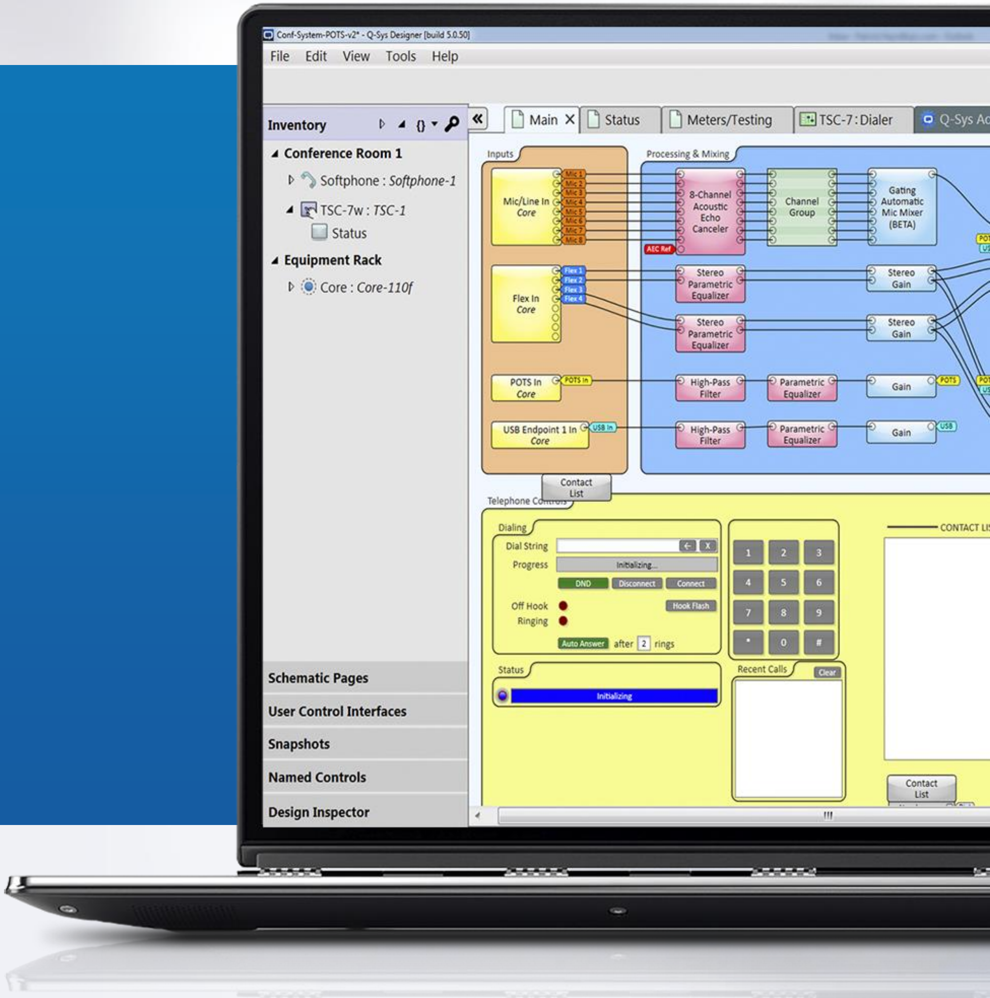
quantum program

learn | install | maintain | succeed

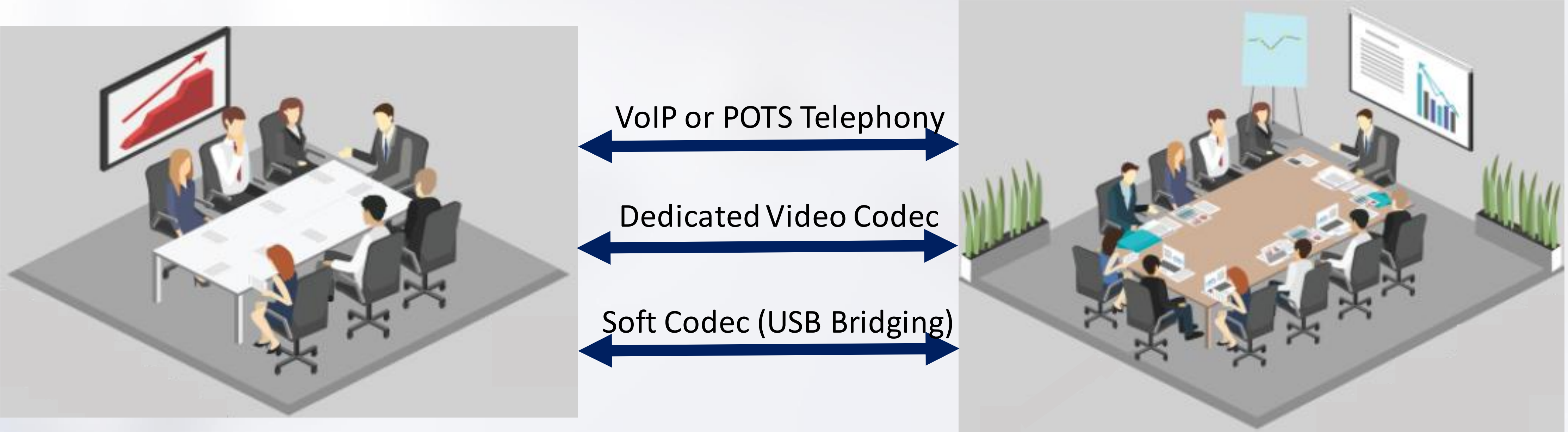
Q-SYS conferencing systems

best practices for systems using AEC



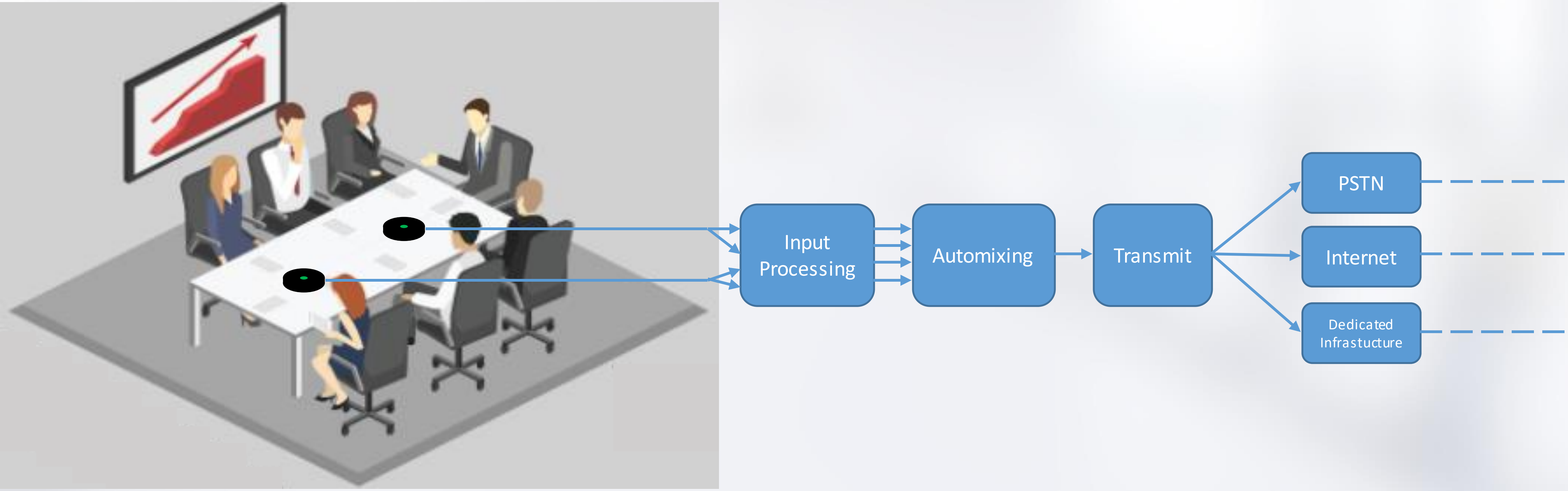
Conferencing Systems

Conferencing systems allow meeting participants in one room to communicate with those in another



Conferencing Systems

Transmit Path
'Far End'



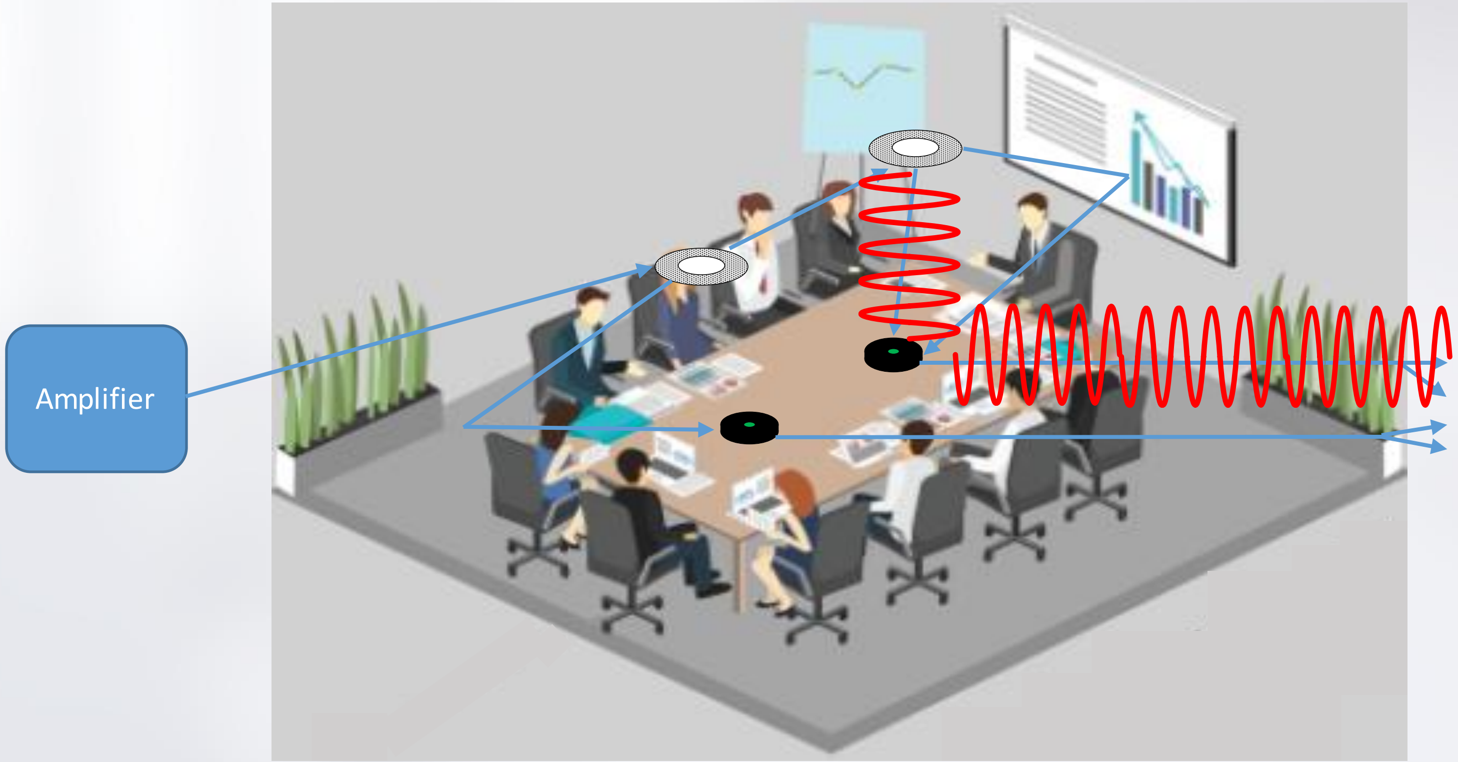
Conferencing Systems

Receive Path

‘Near End’

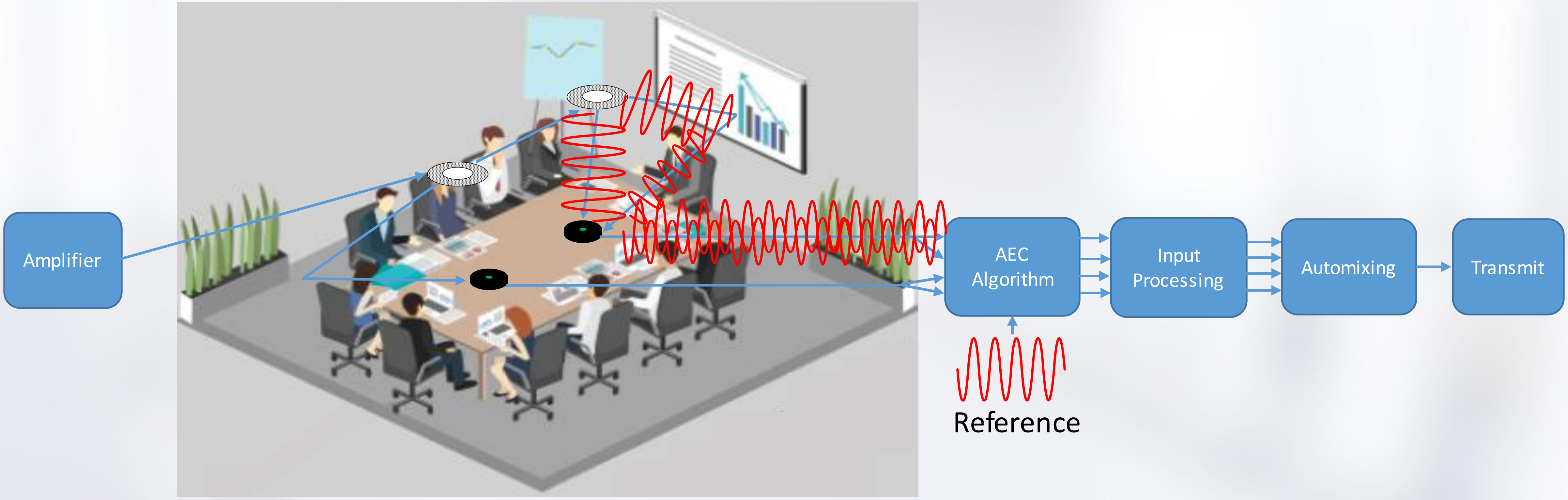
Receiver → Mixing with Local Sources → Output Processing → Amplifier

Acoustical Echo Cancellation



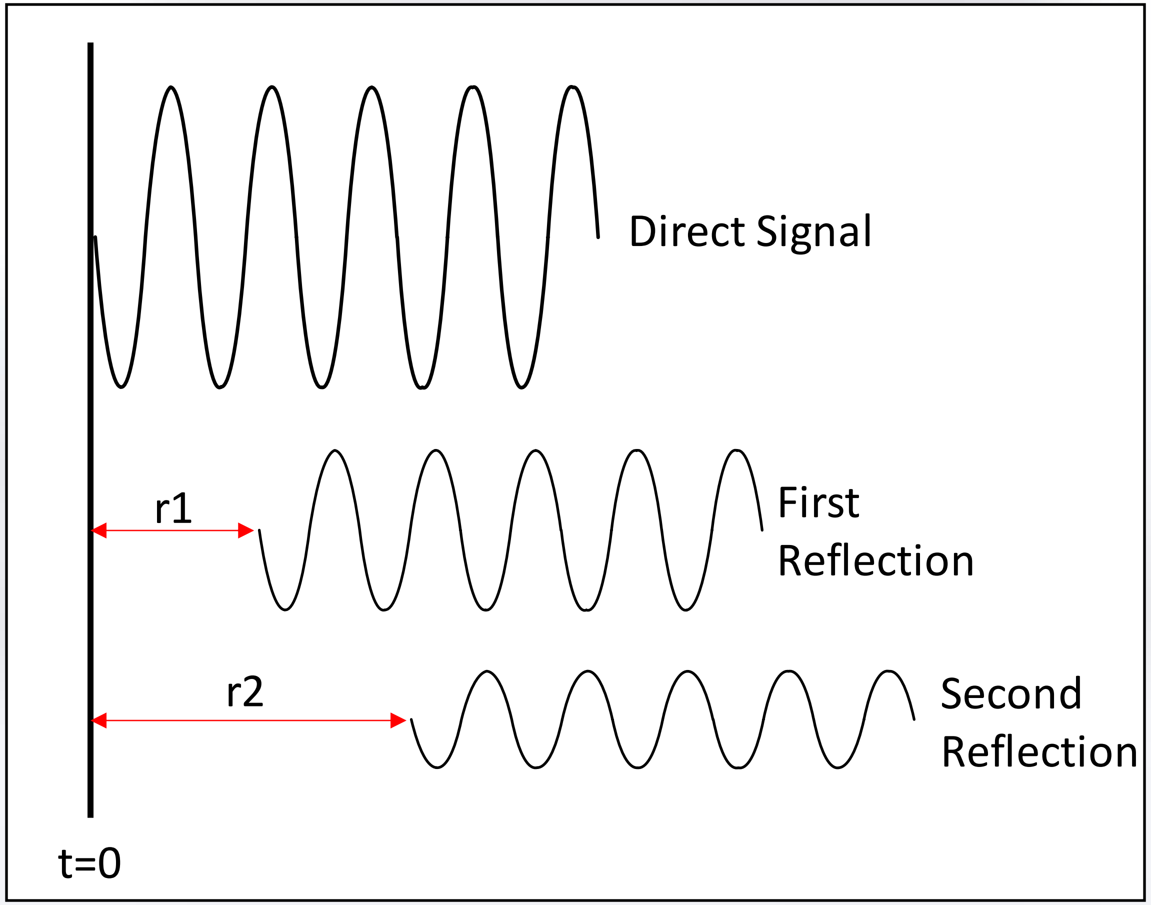
Acoustical Echo Cancellation

Acoustical Echo Cancellation



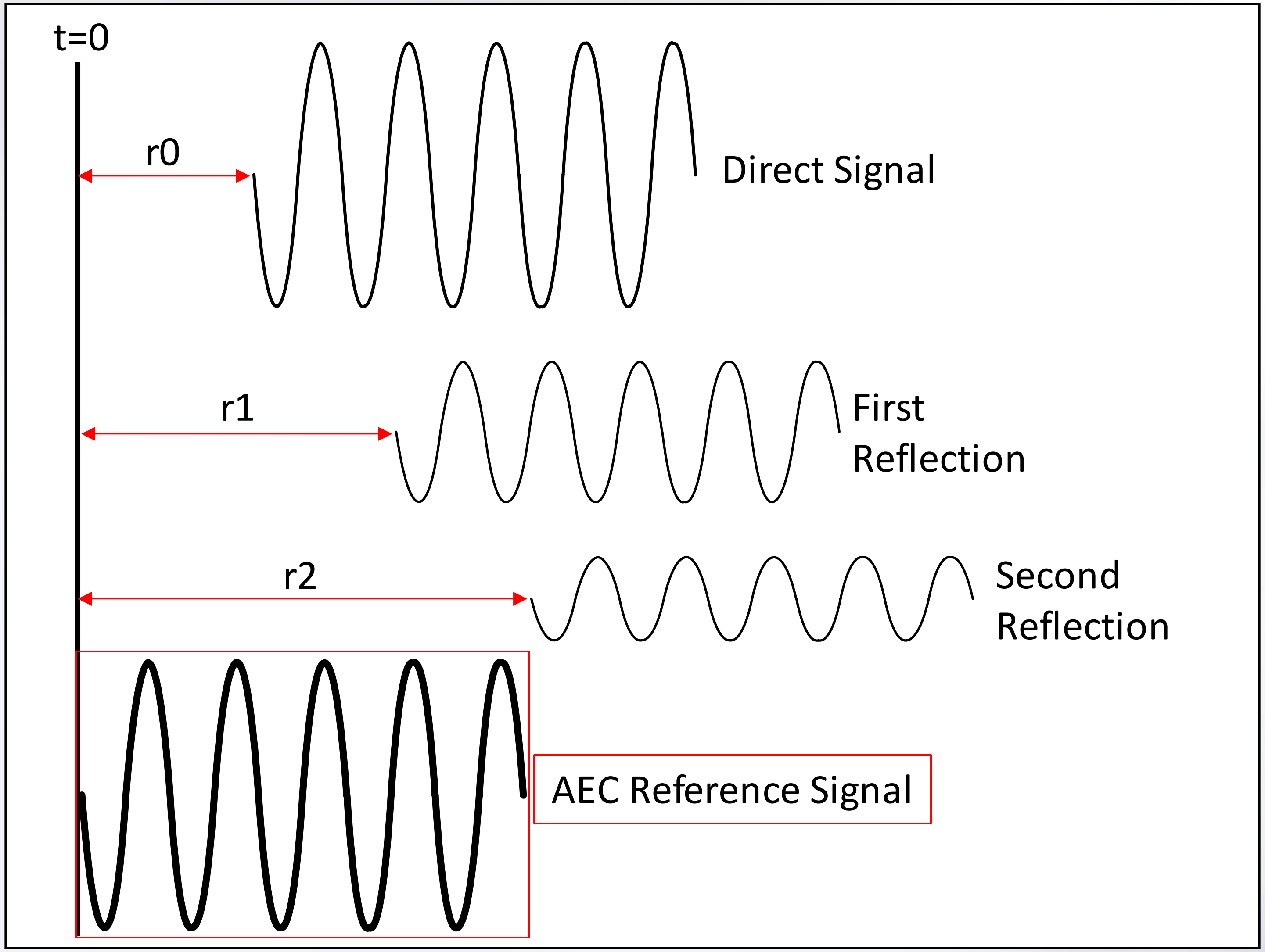
Echo Paths

- At the Microphone



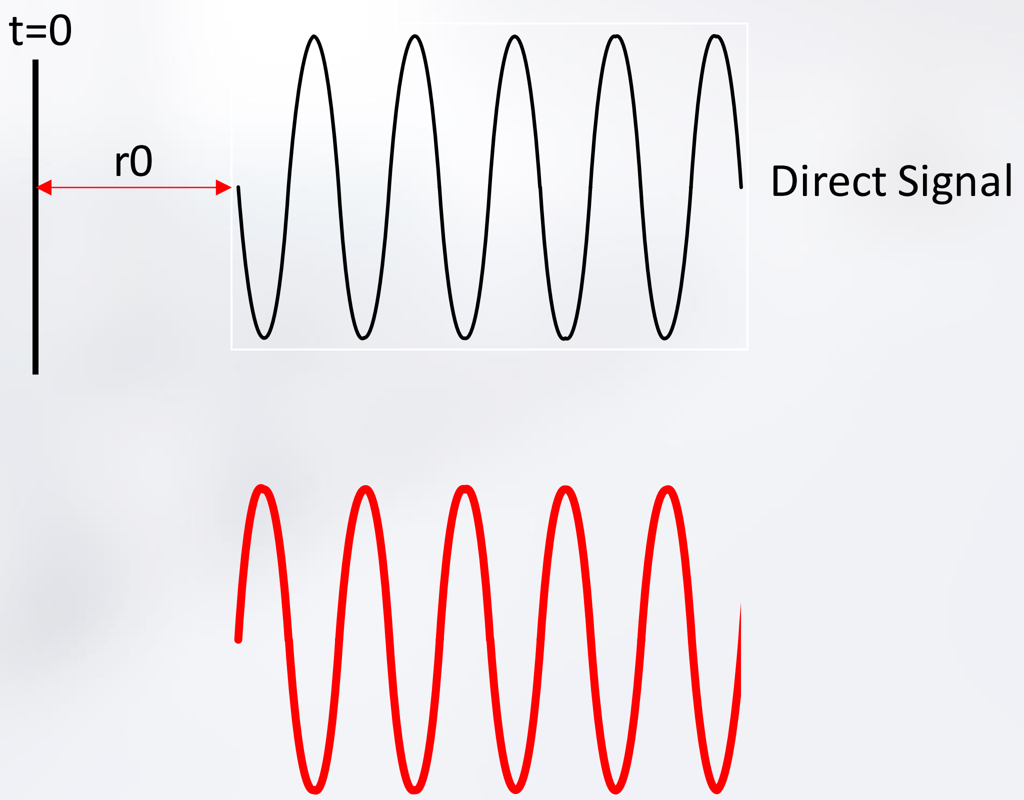
In the Algorithm

- AEC algorithm receives the signal(s) from the mic
- These signals compared to the AEC reference.
- The adaptive filter goes to work!



Adaptive Filter

Scales and time shifts reference signal to eliminate echo



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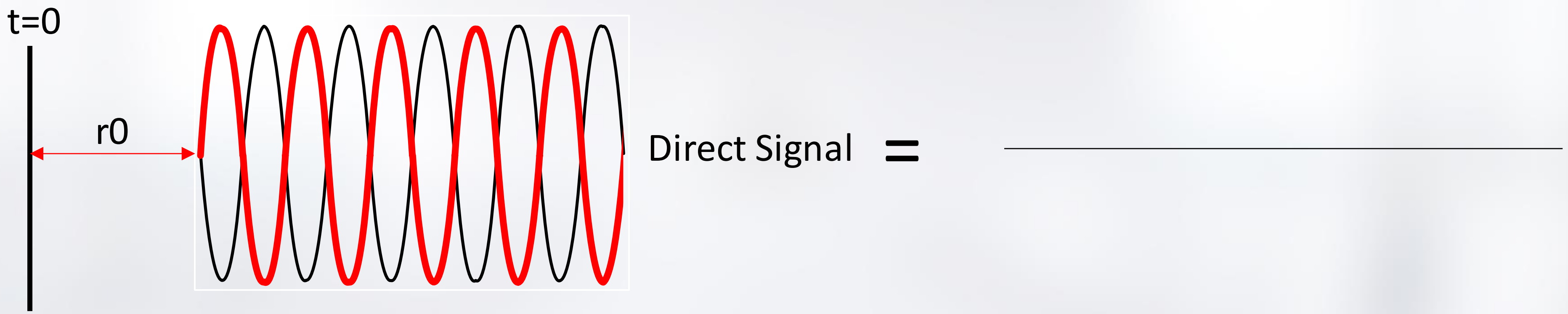

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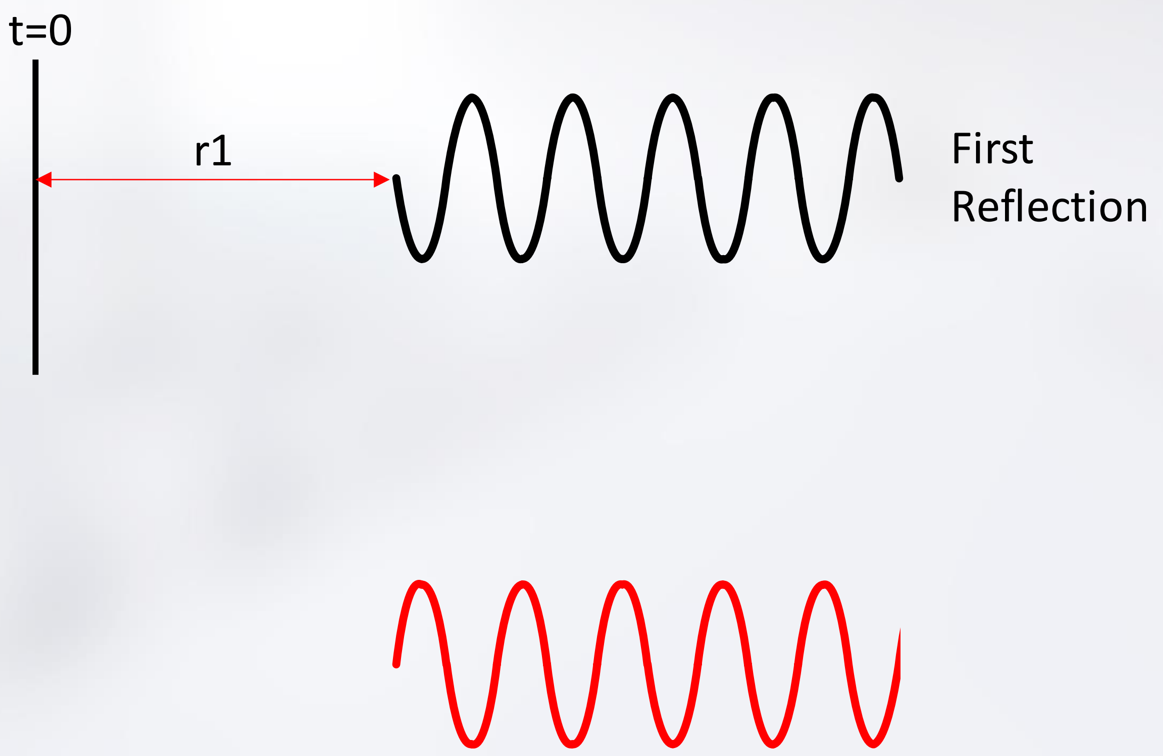

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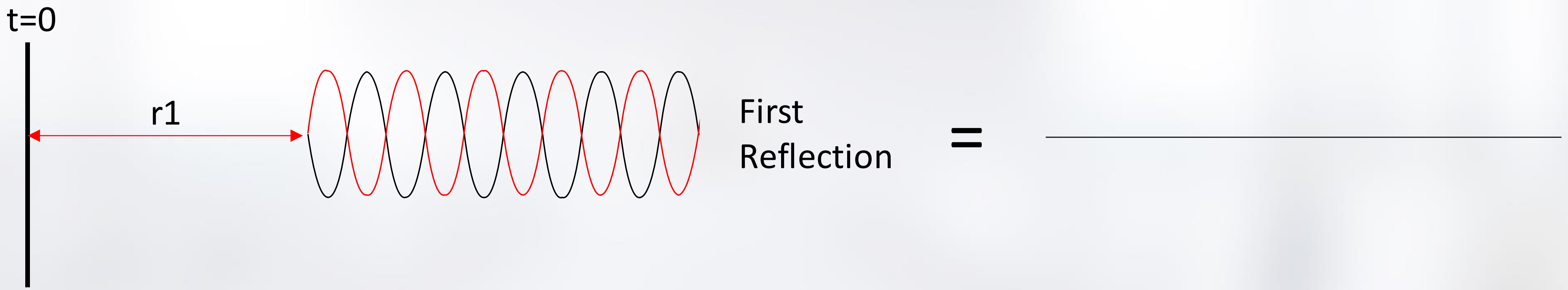

Adaptive Filter

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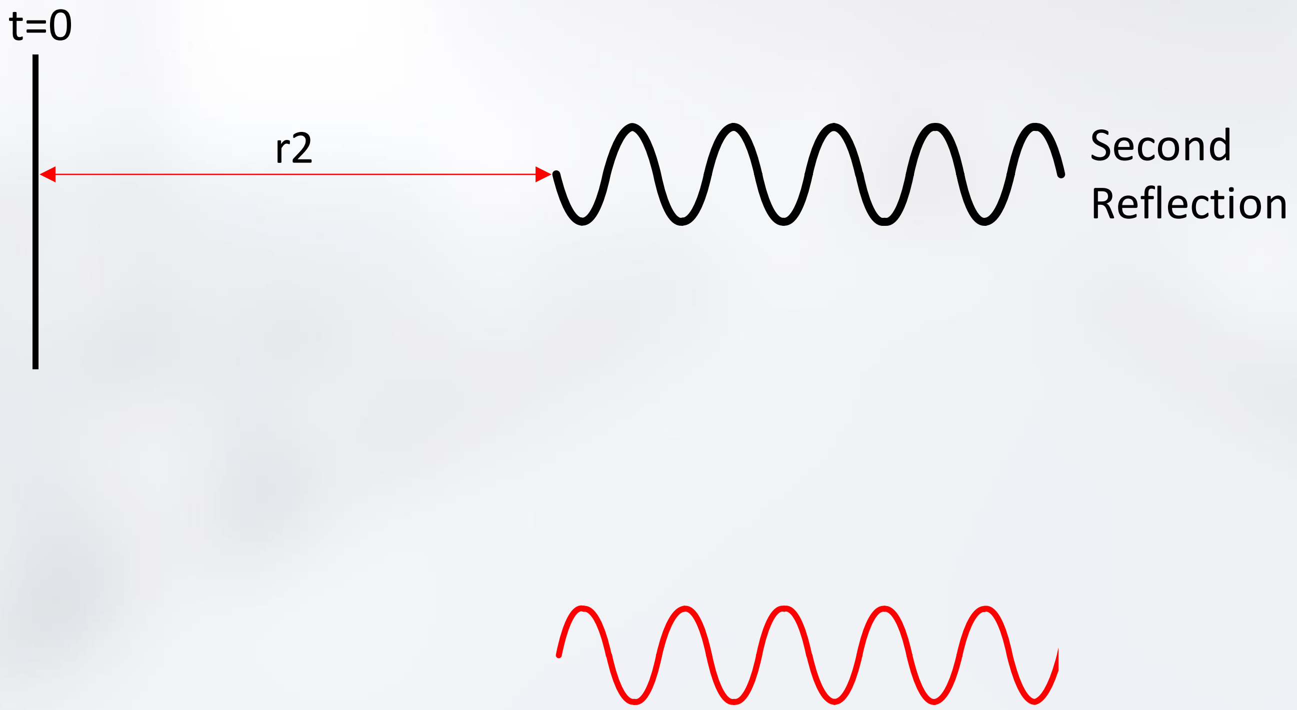

Adaptive Filter

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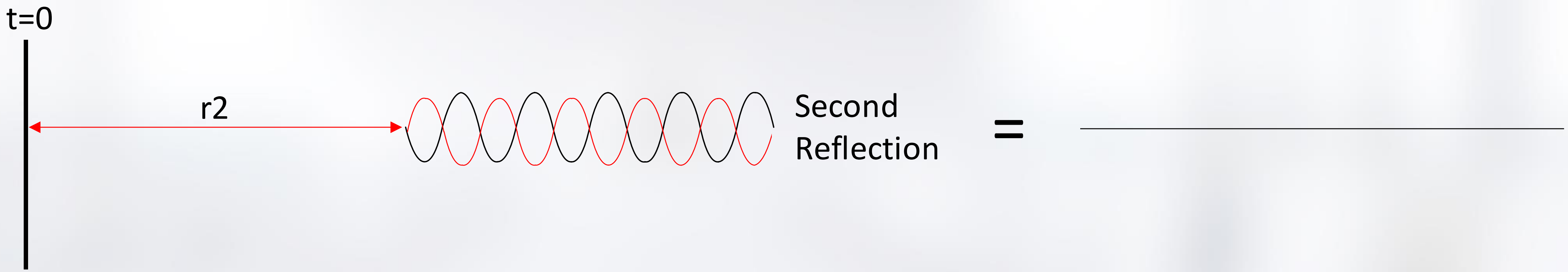
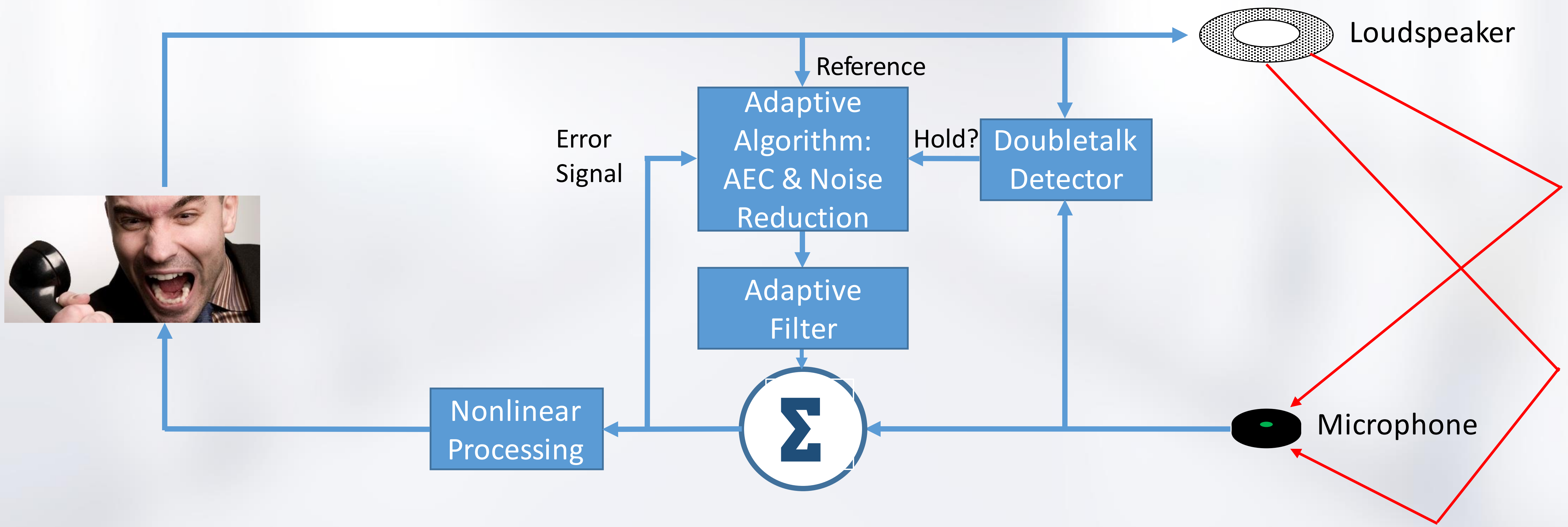


The Algorithm in Detail



Conferencing Terminology

Near end

- The room with AEC to cancel echo to the other parties on the call

Far end

- The room on conference with the near end. The AEC in the near end room cancels the far end echo

AEC Reference

- The signal we wish the AEC algorithm to remove from the incoming mics

Convergence

- Adaptation of the filter to reach successful echo cancellation

Conferencing Terminology

Latency

- It takes awhile to do all this
- Need some ‘lookahead’ time
- ‘Old’ algorithm – 13.3ms
- ‘New’ algorithm – 21.4ms 110f, 10.7ms all other Cores (QSD 6.0.0 and higher)

Tail Length

- The latest possible reflection that can be cancelled
- 100, 200, 300, 400ms options in Q-SYS
- This effectively adjusts the length of the adaptive FIR filter
- The longer the tail length, the more DSP resources consumed

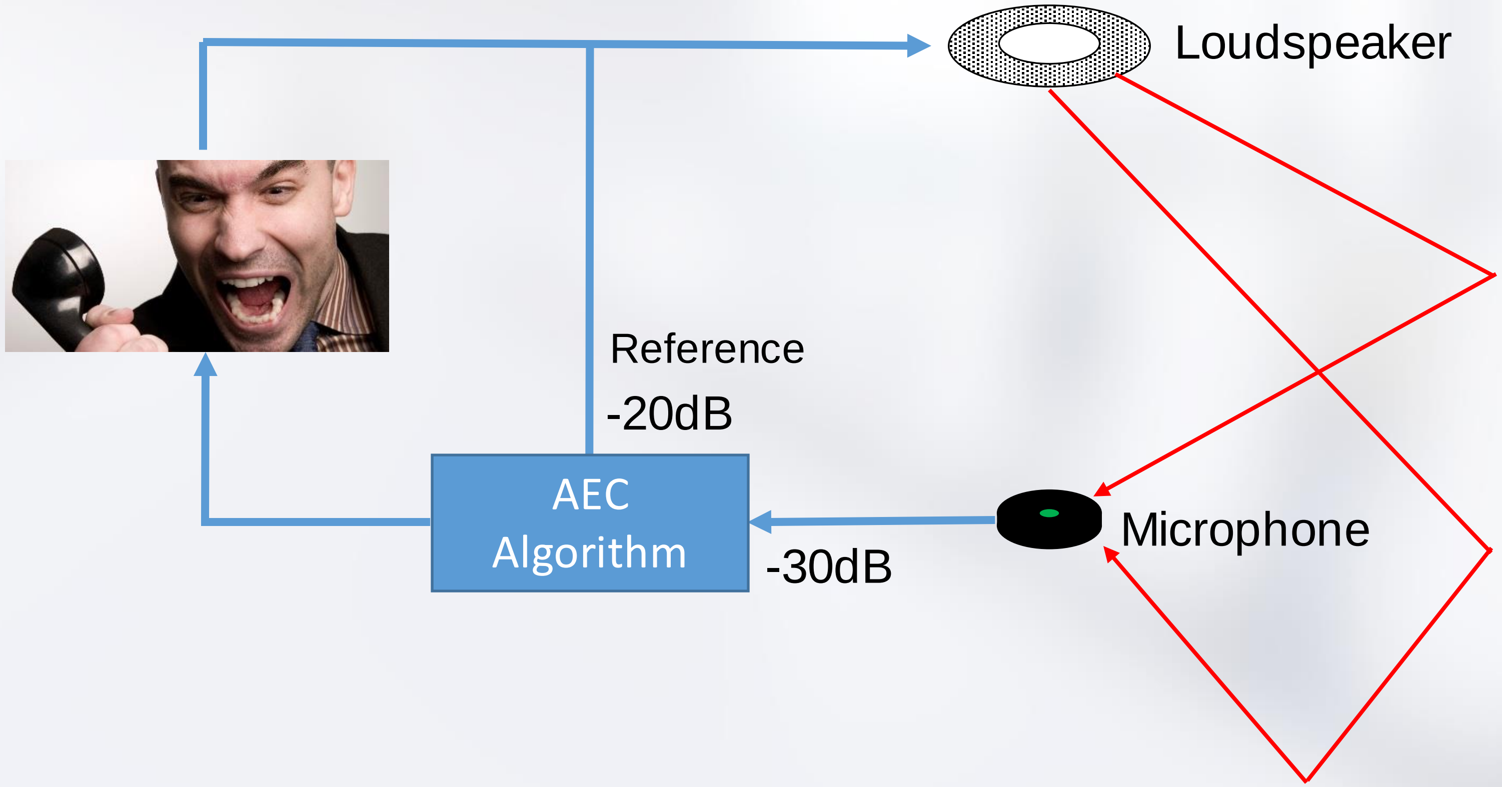
Conferencing Terminology

Reference to Microphone Level Ratio (RMLR)

- The difference in level between the reference signal and the resulting signal at the mic

- RMLR = Lref – Lmic
  - (expressed as loss)

- RMLR = -20dB-(-30dB)=10dB



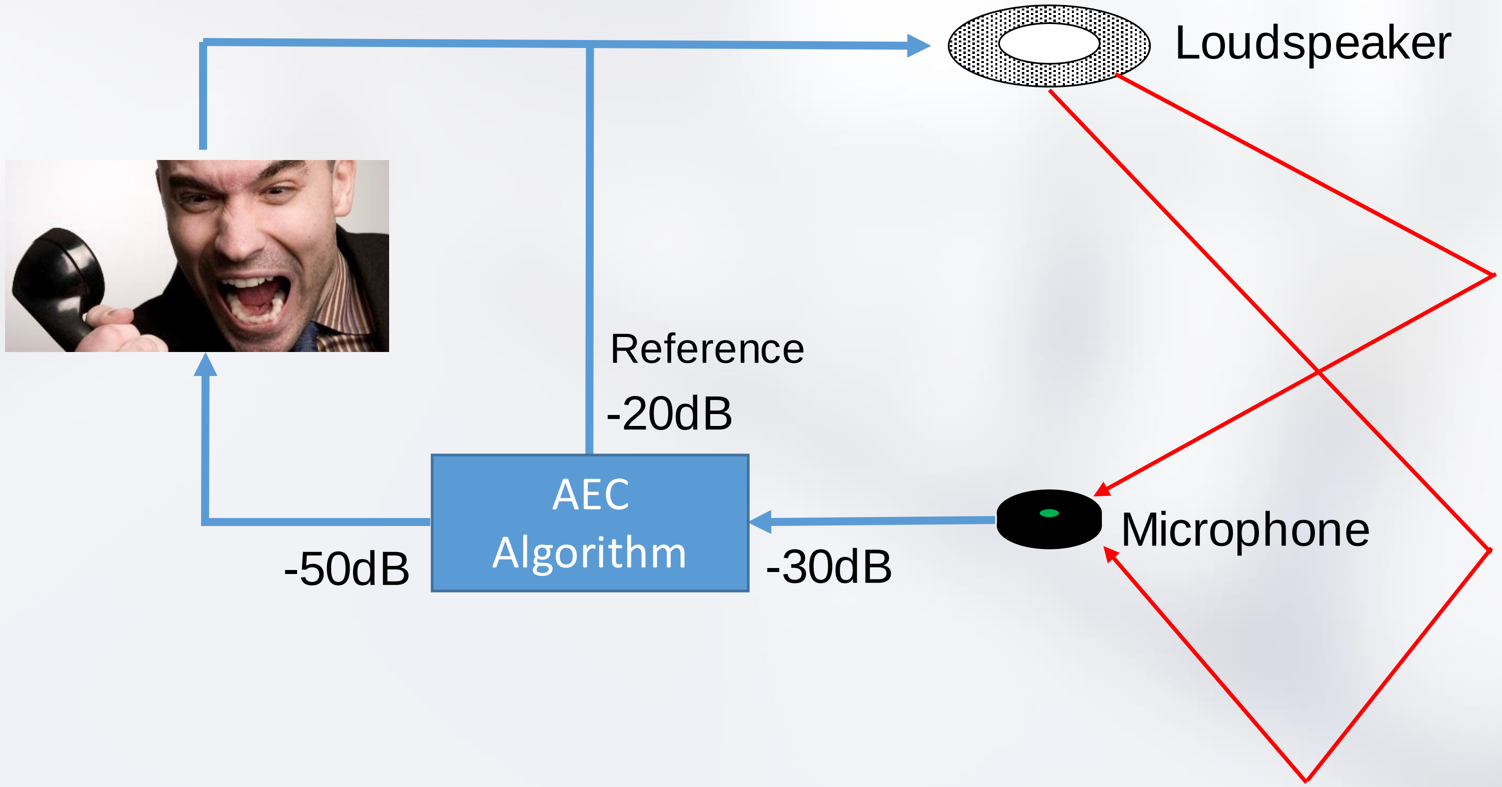
*AKA, Echo Return Loss (ERL)

Echo Return Loss Enhancement (ERLE)

- The difference between the level before and after AEC processing

- ERLE = Lmic – Lpaec
    - (expressed as loss)

- ERLE = -30dB-(-50dB)=20dB



Conferencing Terminology

Residual Echo Suppression

- Same as Non Linear Processing
- Removes error signal after AEC

Voice lift

In large rooms where the near end microphones are sent to the speakers in the near end

Mix Minus

In larger rooms where there are a number of distinct speaker circuits for voicelift purposes. Microphone signals are strategically mixed to each output to maximize gain before feedback and retain the most natural sound

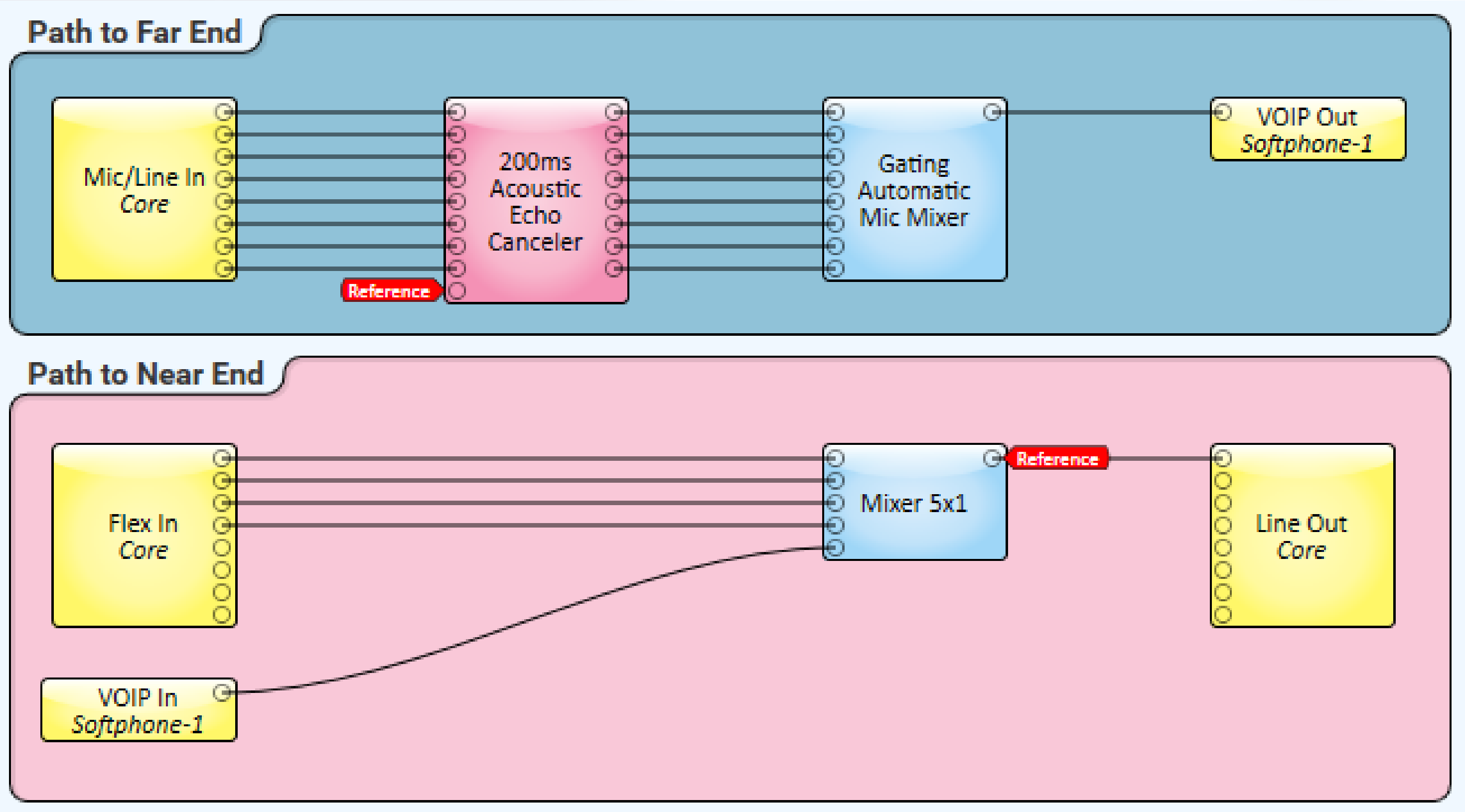
Q-SYS Conferencing Signal Flow

Path to far end

- Local microphones
- AEC
- Automixing
- Output to conference

Path to near end

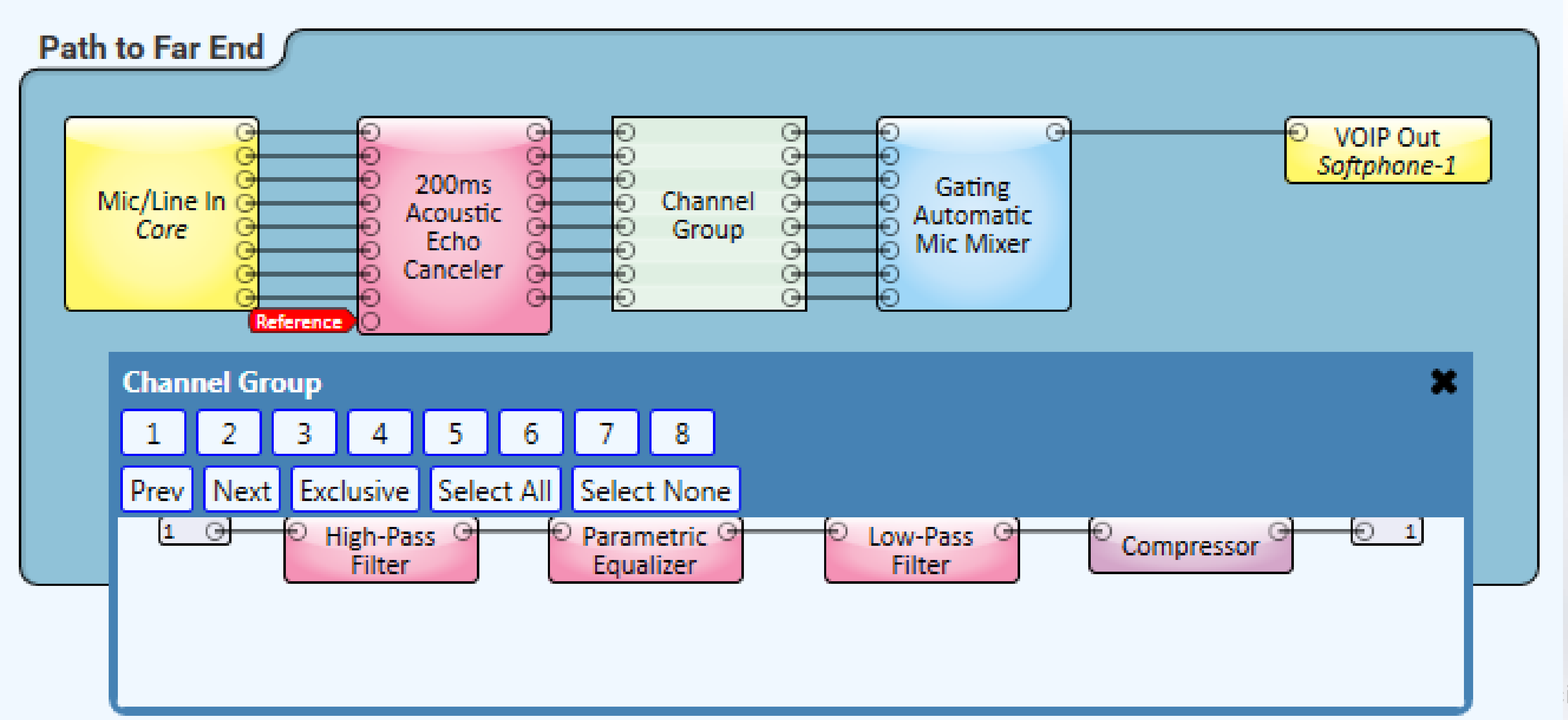
- Program sources
- Far end conferencing



Microphone Processing

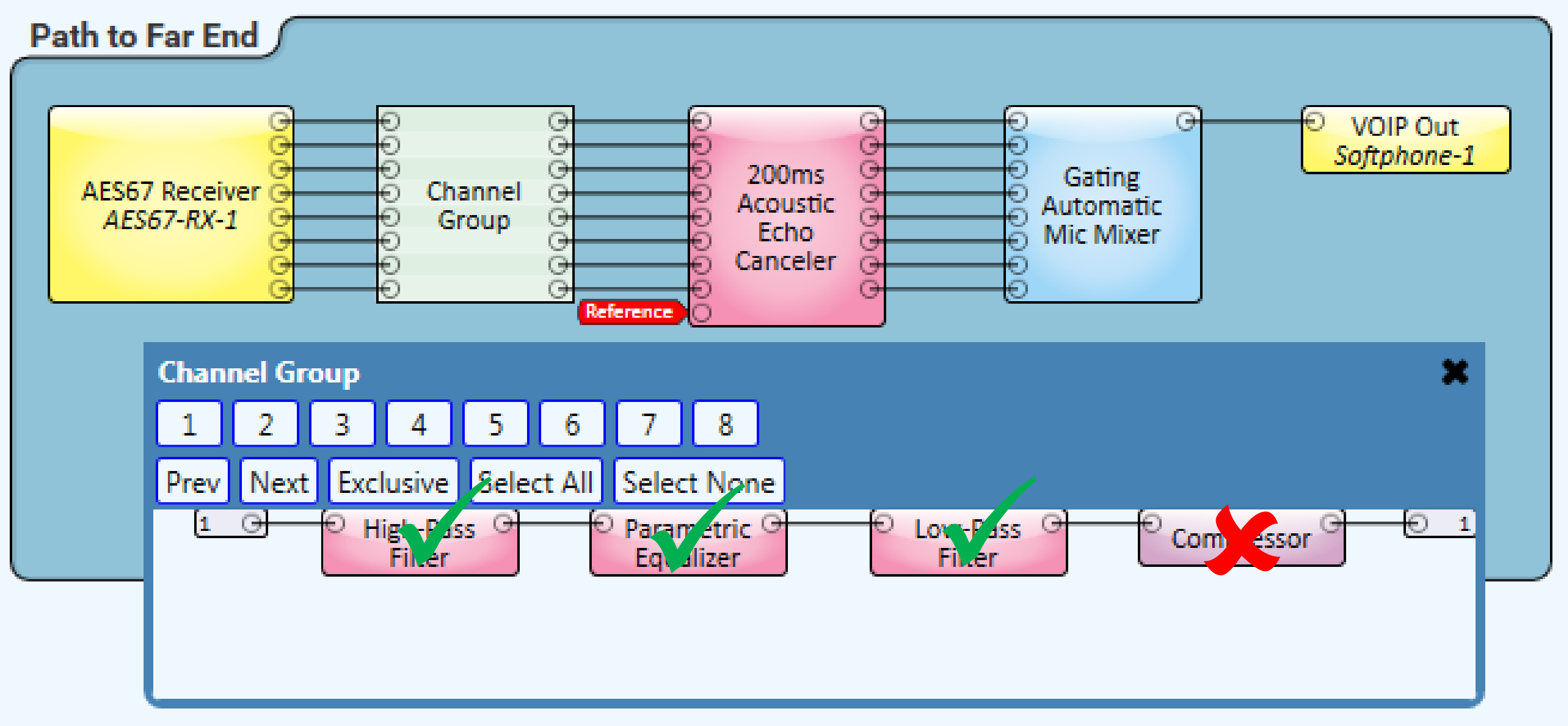
Best Practice

Apply processing <u>after</u> AEC



Microphone Processing

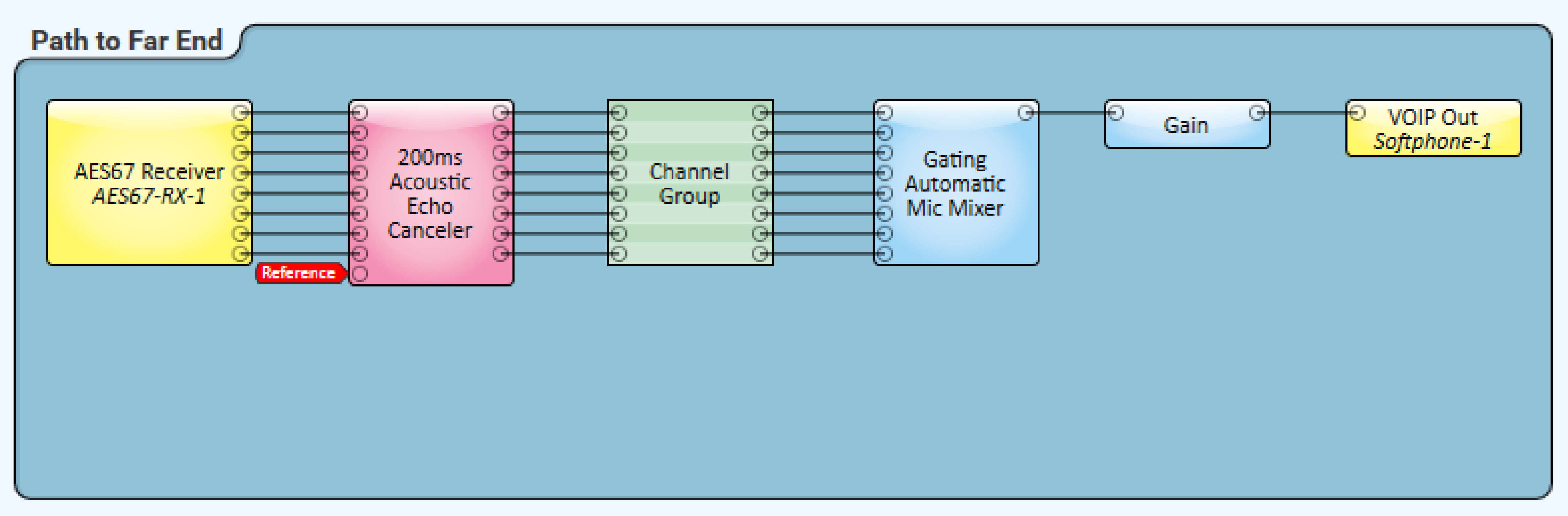
- In some cases it is acceptable to apply EQ ahead of AEC
- NEVER place dynamics or non-linear objects ahead of AEC



Microphone Processing

Recommend unity gain throughout the signal chain

- If required, place user transmit gain ahead of transmit block
- Set gain control range to avoid clipping



AEC Reference

The most important factor in successful AEC is having the **correct reference signal**

When no voicelift the adage is

- "What goes to the speakers goes to the reference"
- ALL far ends
- Program sources (especially if they're being routed to the far end)

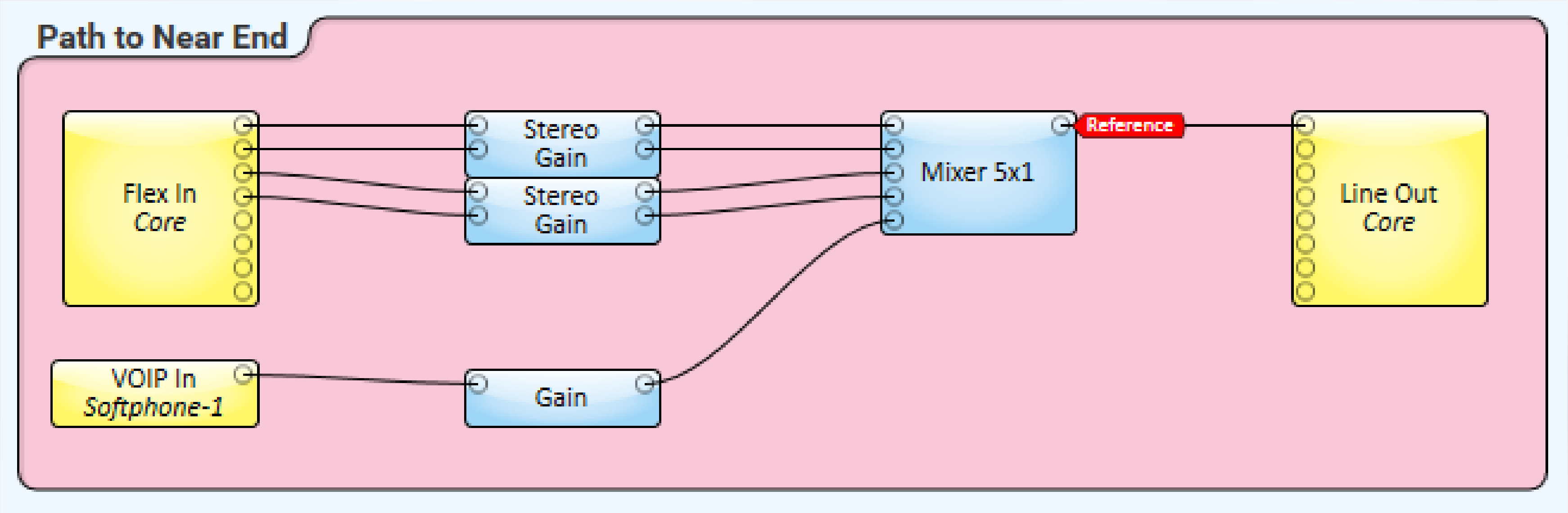
NEVER

- Put a mic signal into its own reference

Near End Signal Path

Place user gain controls ahead of mixer

- Want level in room and reference level to change at the same time
- Avoid overall output level controls where possible
- Set gain control limits to avoid clipping

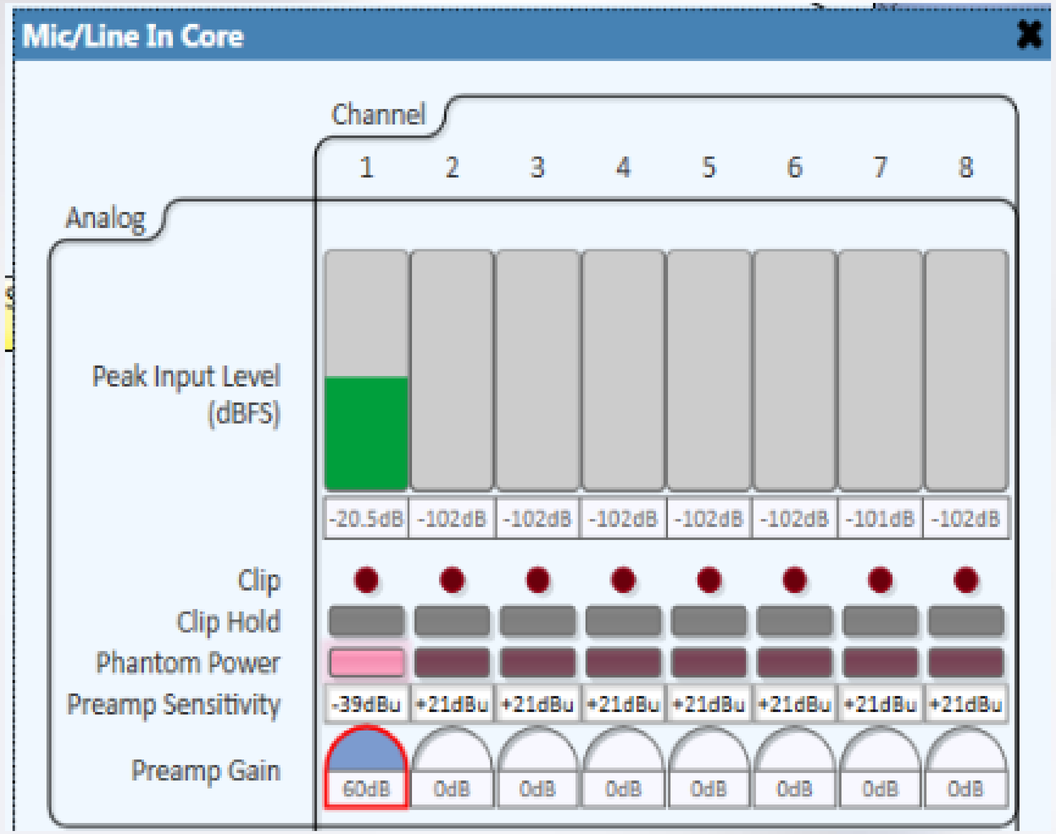


Conferencing System Setup

1. Use preamp 'analog' gain to bring microphone signals to -20dBFS nominal

(Yes, this applies to AES67/Dante Mics as well)

These systems should be calibrated to send nominal level with no adjustments in Q-SYS

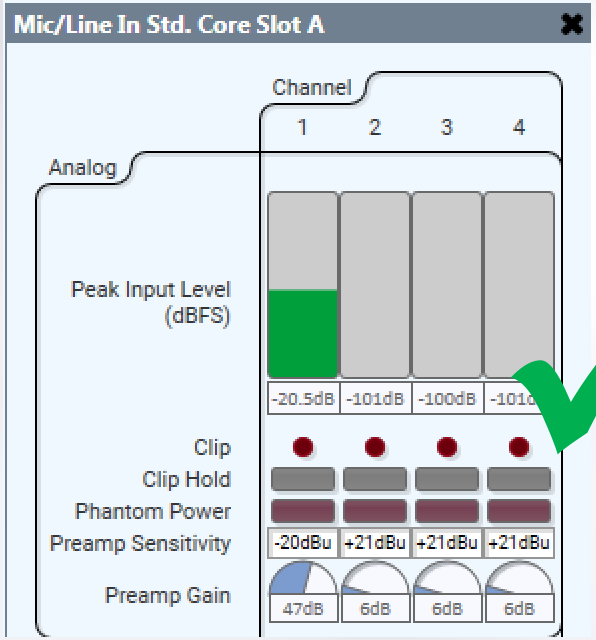


Conferencing System Setup

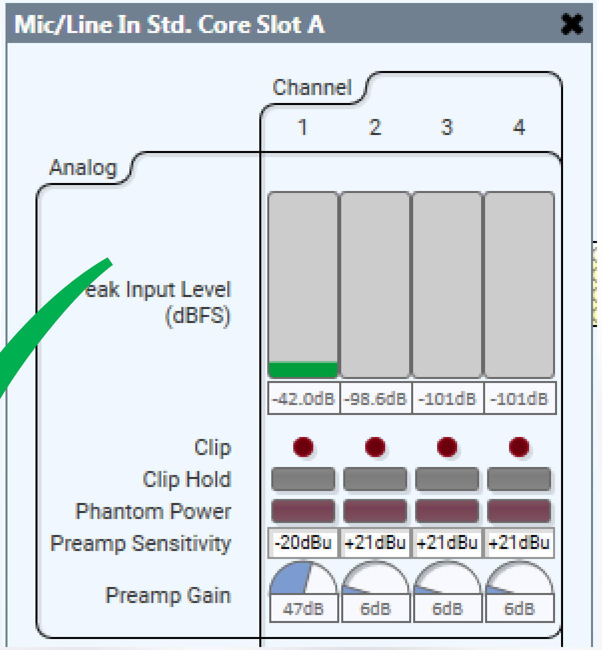
2. Check Microphone Signal to Noise Ratio (SNR)

If SNR<15dB, intelligibility standards are not met, noise reduction algorithm cannot account for this

If SNR standards met, leave mic signals at unity gain through complete signal chain



Microphone Nominal



Microphone Noise Floor

Conferencing System Setup

3. Set amplifiers to minimum level <u>OR</u>

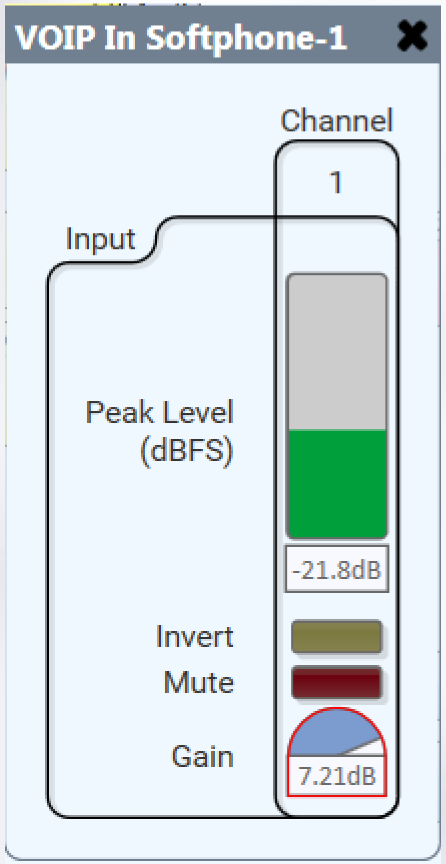
Set output block 'Max RMS' gain to -40dBu (if no controls on amplifier)

- Use preamp 'analog' gain to bring all program source inputs to -20dBFS nominal
- Leave program sources at unity gain through signal chain
- With calibrated program source, slowly bring up amplifier or 'Max RMS' setting to comfortable listening level (typical 70dB SPL)

Conferencing System Setup

4. Make test call

- Adjust far end(s) gain to nominal receive level of -20dBFS
- With listener at far end(s), AEC can be fine-tuned

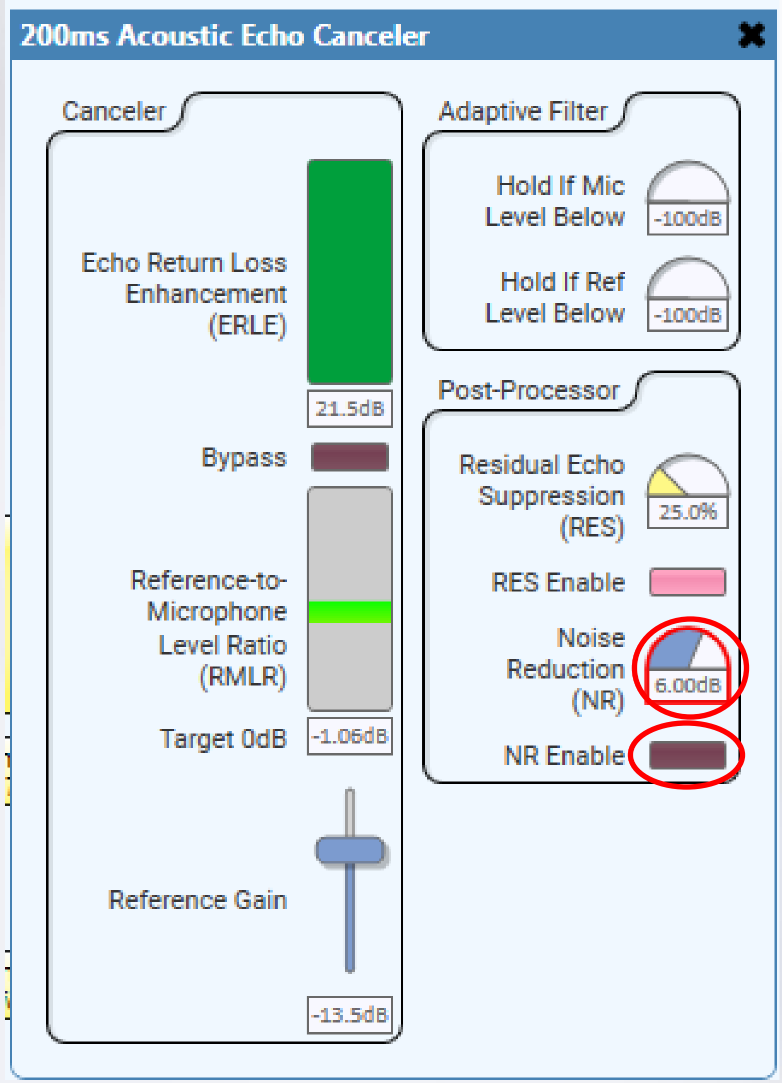


Conferencing System Setup

NR Enable/Level

NR reduces only steady state noise

It can color microphone signal slightly as more NR is applied



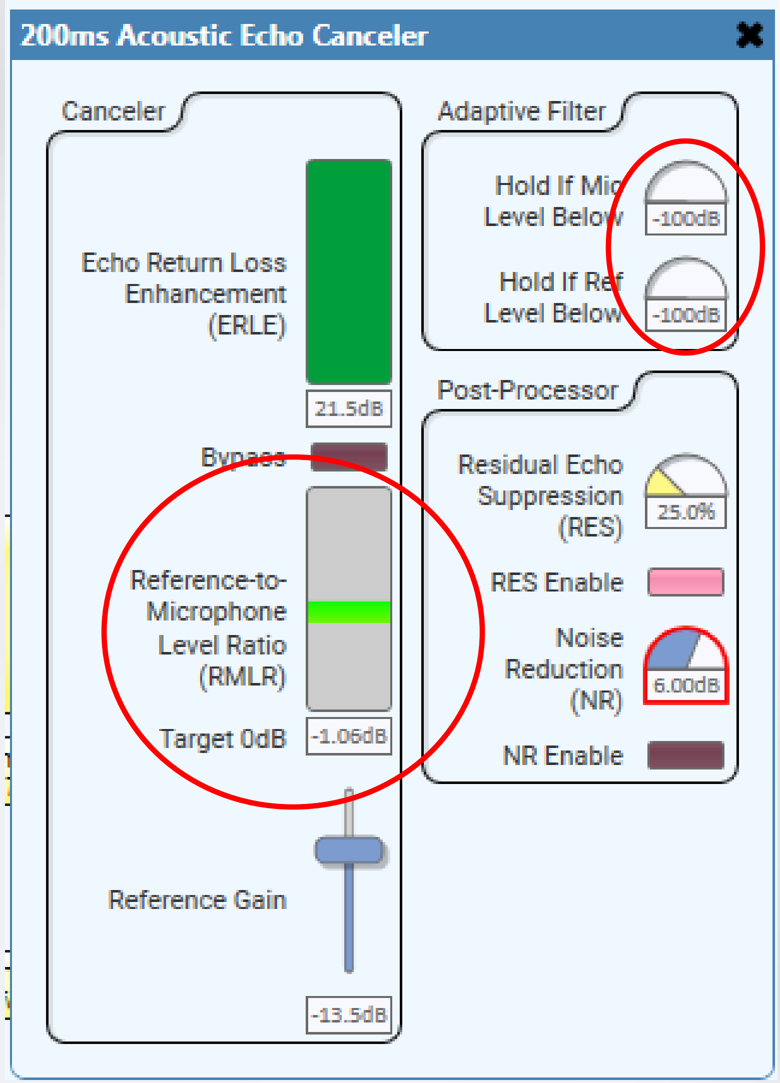
Conferencing System Setup

RMLR

- Verify the meter is green when the far end speaks
- Can adjust reference gain if needed

Adaptive Filter

- Holds convergence below threshold (for mics that are muted ahead of the AEC algorithm)
- Set a few dB below noise floor when unmuted
- Can leave at -100dB if mics only muted after AEC algorithm



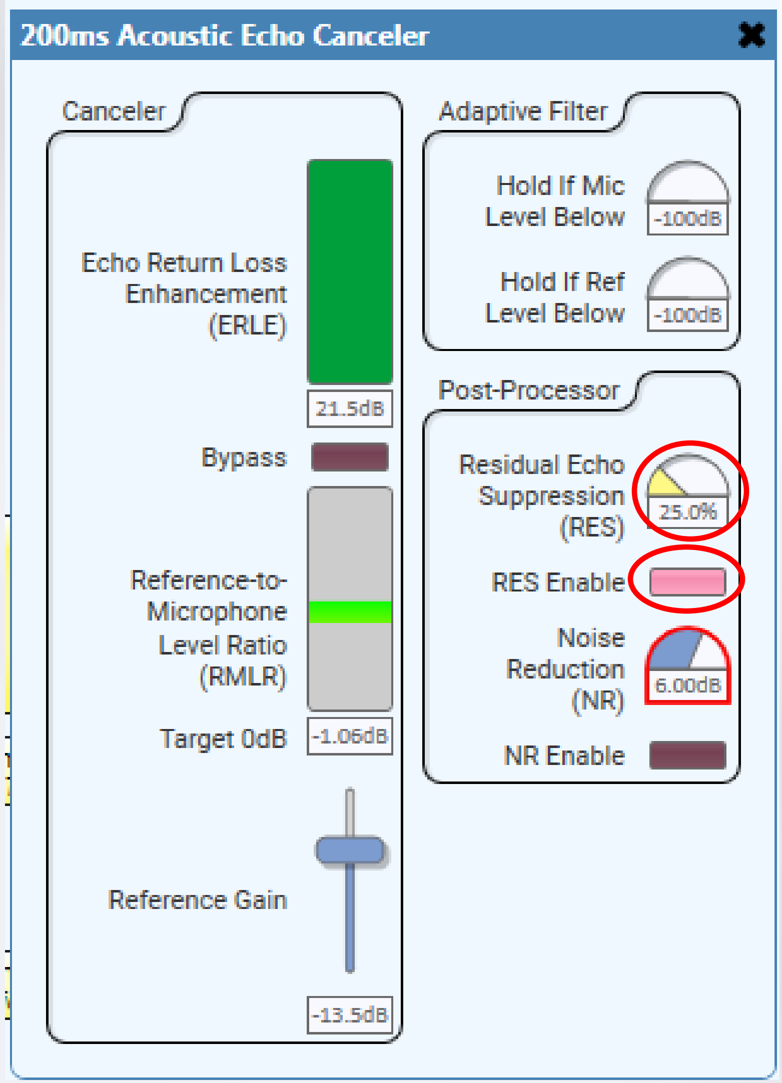
Conferencing System Setup

RES Enable/Level

- This sets non-linear processing
- Increase if there is residual echo
- Will approach 'half-duplex' as this is increased

Test double-talk condition here

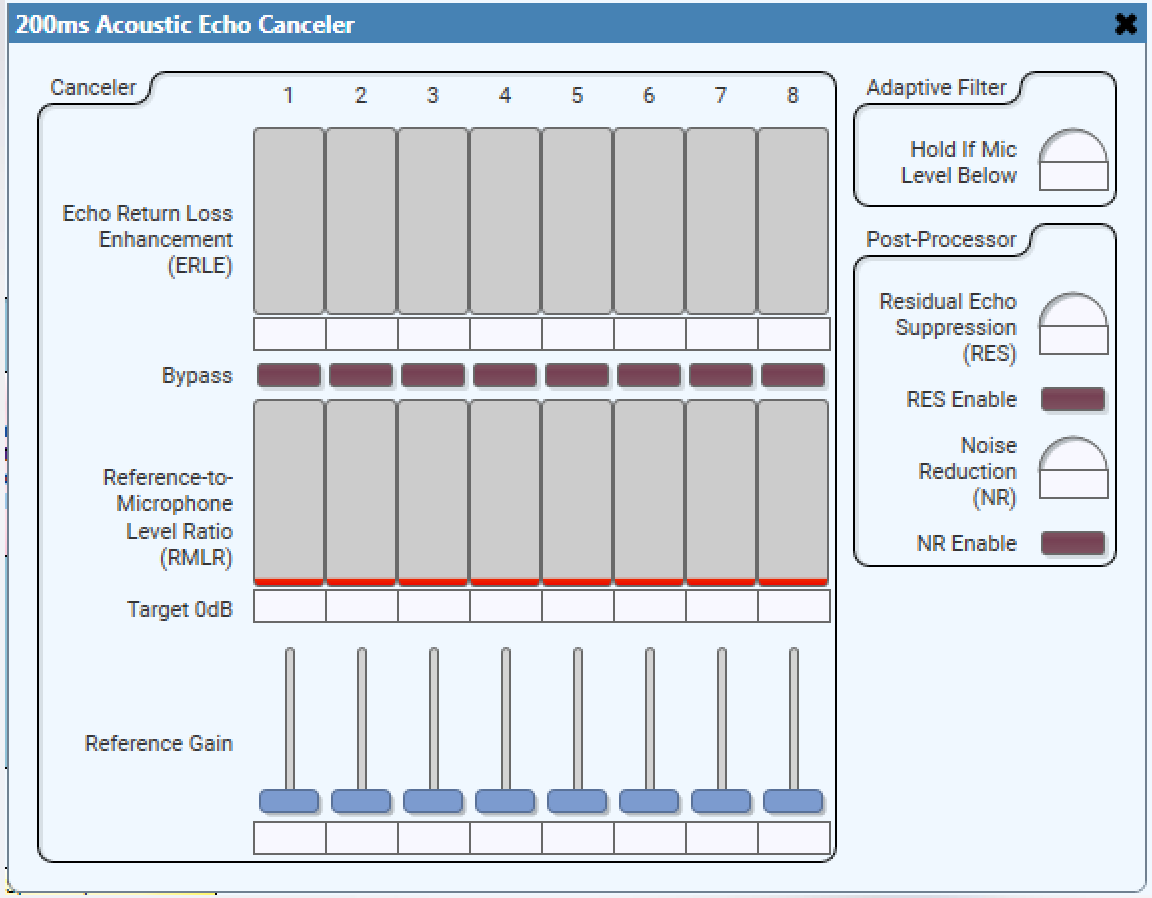
- Increase RES to get desired performance



Conferencing System Setup

5. Repeat step 4 for each conferencing type

   - Then finally a test call with all conferencing types simultaneously



Conferencing System Setup

What if there's still significant echo?

- Mute microphones to verify that it is really echo and not direct coupling
- Make sure signals aren't misrouted or looped
- If 'echo' still there, then mute inputs until you identify misrouted source

Conferencing System Setup

What if there's still significant echo?

- Check AEC tail length
  - If at 100ms, can extend to 200ms
  - If at 200ms, not likely to improve by increasing to 300/400ms
  - Only very large rooms will require 300/400ms tail length

- Recommend methods to reduce reflectivity of room

Questions/Discussion?